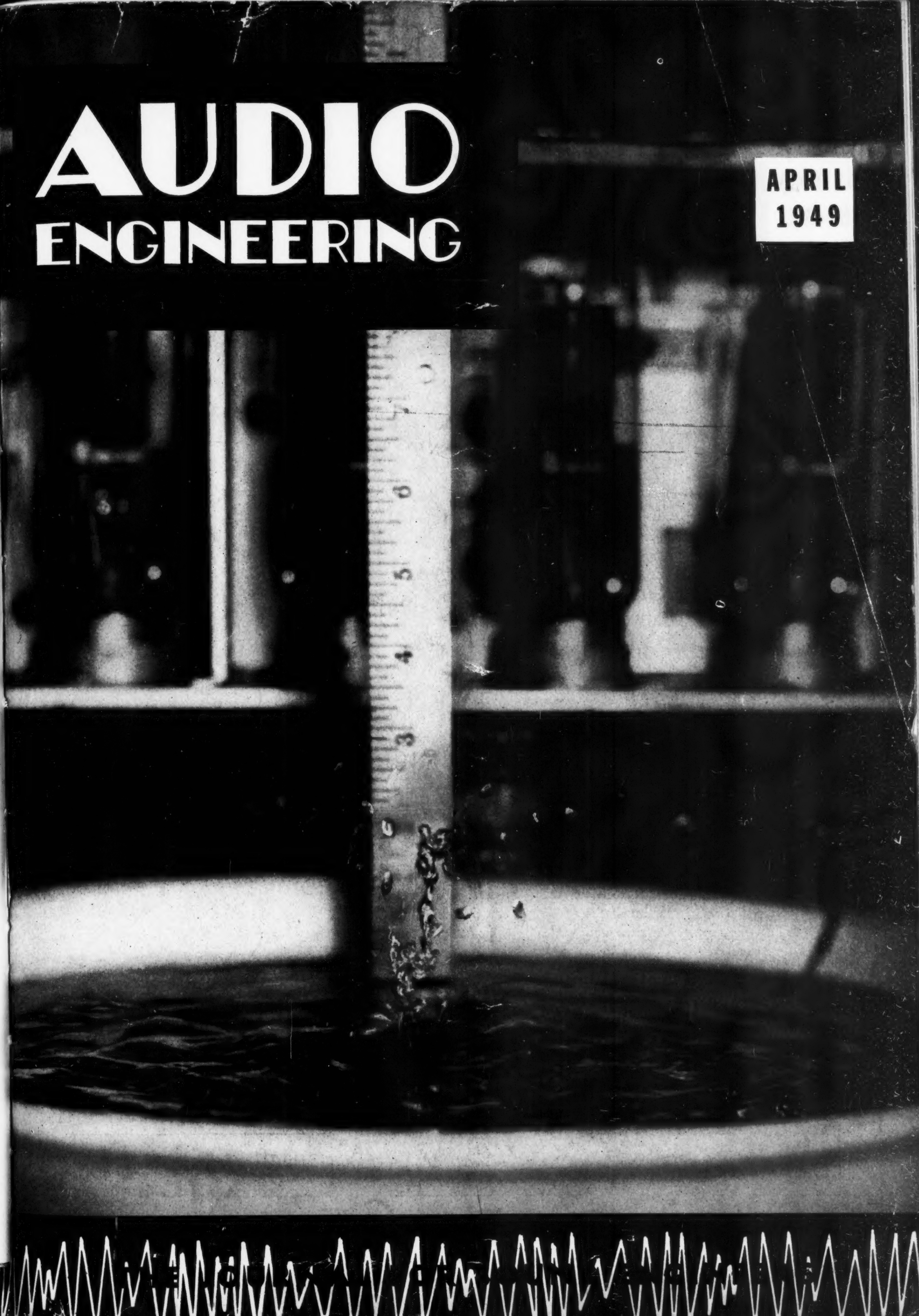
AUDIO ENGINEERING

APRIL 1949



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COVER

Ultrasonic fountain in transformer oil, created by an immersed quartz crystal excited by 150 watts of energy at 400 kc. Converted BC-375, surplus transmitter employed as generator, is seen in the background. Photo by Lewis S. Goodfriend.

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John H. Potts

1892-1949

It is with profound regret that this page—which for every issue since the inception of this publication has brought to you the Editor's Report—must be devoted this month to chronicling the untimely passing of John H. Potts, co-founder, editor, and publisher of AUDIO ENGINEERING.

John Potts, one of the early pioneers in the development and progress of the electronic arts, devoted his entire life to our industry as editor of some of its leading publications, as author of innumerable articles, and in engineering capacities with the Radio Corporation of America, the General Electric Company, Westinghouse Electric Corporation, and the Sperry Gyroscope Company.

Mr. Potts was president of Radio Magazines, Inc., and editor of *CQ* and *Radio*. The latter was succeeded in May, 1947, by this publication, and on December 1, 1948, Mr. Potts became publisher of both *CQ* and AUDIO ENGINEERING.

John H. Potts, born November 8, 1892, died March 16, 1949. Author, editor, publisher, engineer. R. I. P.

— *Letters* —

Tone Controls

Sir:

In reading over the Feb. issue, I was no little amused by one of the letters. Speaking of wordage wandering and confused implied thinking—Take "Does the fidelity rating go higher directly as the number of controls?" Obviously they have one purpose only . . . to give the listener his own tonal balance choice. If he were concerned with just one source of program material or better still *one* record, he would need no controls whatsoever . . . not even an On-Off switch . . . he could just pull the plug!

If the listener could take his own recorder to a concert for example, secure his own balance and handle its processing all the way down, his task would also be much simpler. However, when the many music sources are considered, AM, FM, different size and quality ET's and records are made by twenty or thirty different manufacturers who use two or three different cross-over frequencies and recording characteristics, these controls come in handy!

From Letters in the Oct. issue "Let your musical sense determine the curve and your engineering its ultimate reproduction result," let us pose another problem to further confuse the thinking! The musician or the average ear or better attends the same concert with a person who has a "tin ear." Same seat, same music, same everything except ears and they will come away with different impressions of that concert. Suppose their ear impressions were converted exactly to a record and they both build amplifiers, etc. and accurately recreate their impressions . . . which is high fidelity?

Absolute tonal balance . . . what is the standard? Whose ear is the criterion? The conductor has his own balance and each listener has his. If the two coincide, then the whole performance will be more pleasing. Pitch? . . . Orchestras in different countries tune up to different pitches . . . they are not standard as yet!

All this is so well known that I might as well sign off and stop annoying people!

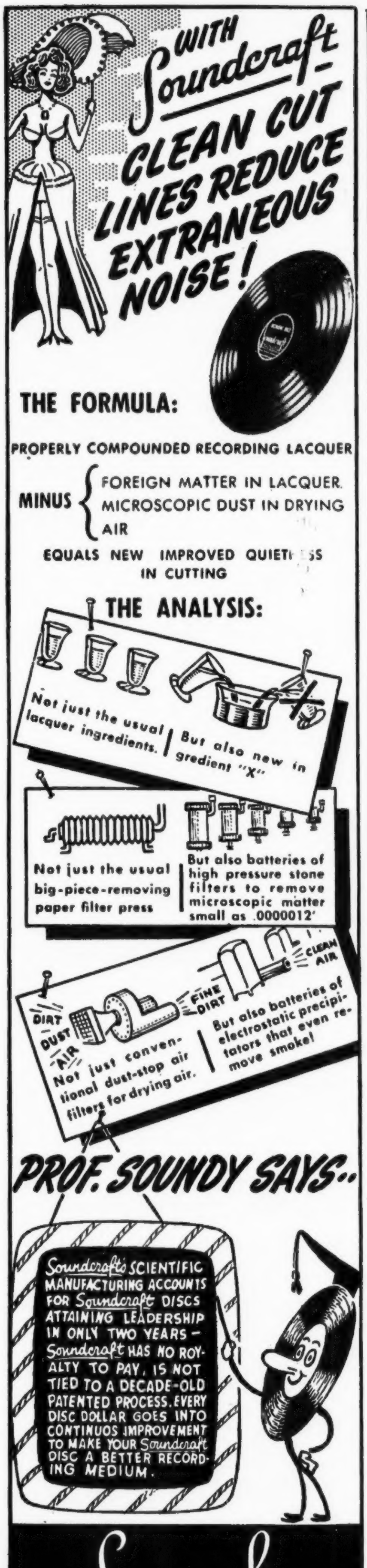
J. P. Cook
2609 Buena Vista
Bakersfield, Calif.

LP Reactions

Sir:

Canby has scored a triumph in his March column! In his discussion of long-playing records he has hit the nail right square on the head—and

[*Continued on page 46*]



REDUCING EXTRANEOUS NOISE IN RECORDING

By A. C. Travis, Jr.*

Discussing disc recording with experts in broadcasting and sound studios off and on for ten years brings one inevitably to the point of trying to generalize the recording engineer's problem. Inadequate as making statistics out of memory may be, so overwhelming is the impression of unanimity that a simple summary promptly springs to mind. Regardless of what high-fidelity ambitions may haunt the recording engineer, his soul-searing fear is simply that of making a sub-standard recording of irreplaceable material.

The commonest single cause of sub-broadcasting-standard disc recordings is extraneous noise. So complex are the reasons and cures for this destroyer of otherwise good broadcasting material that they transcend the possible scope of a short article. To oversimplify, however, it is noteworthy that recorded extraneous noises behave somewhat like breakfast foods. They may hiss, swish, crackle, or pop. Since even these few categories of noise cover a lot of Puffed Rice, space requirements hold us down to a limited discussion of "hiss".

The blame for excessive hiss level in disc recordings is generally shared by the blank record and the sapphire stylus. At this point buck passing reaches championship proportions. Most often, however, neither suspect is ever definitely exonerated. The trouble simply disappears by itself. The history of recording disc manufacture, of course, allows little doubt of the fact that with some brands "grey cutting" discs crop up unpredictably from time to time. It is also an admitted fact that sapphire styli may vary so greatly as to make up to 12 db difference in surface noise level. Such variation, while unintentional, usually occurs where low prices dictate loose microscopic tolerances in sapphires.

Some of the more tricky causes of "hiss" include cutting cold discs fresh from the delivery truck, allowing smog (fog-borne soot & dirt) to settle in the grooves, and misalignment of the cutting stylus. Nitrate-coated discs (so-called "acetates") seldom cut quietly unless the aluminum bases are at a temperature between 70° and 90°. Fine or coarse airborne dirt, moisture, or damp dust can spoil the polishing action of the best stylus. Stylus misalignments to be avoided include more than a degree or two off vertical and twist of the shank in installing the stylus in the cutting head.

Today, except for occasional tricky recording problems, the most nervous engineer can fortunately forget his worries. With the new, constantly-improved Reeves Soundcraft discs and Soundcraft styli combining to keep extraneous noise 55 to 65 db below peak signal, it's mighty hard to muff a recording. Soundcraft products have indeed established disc recording anew on a standardized *predictable* basis.

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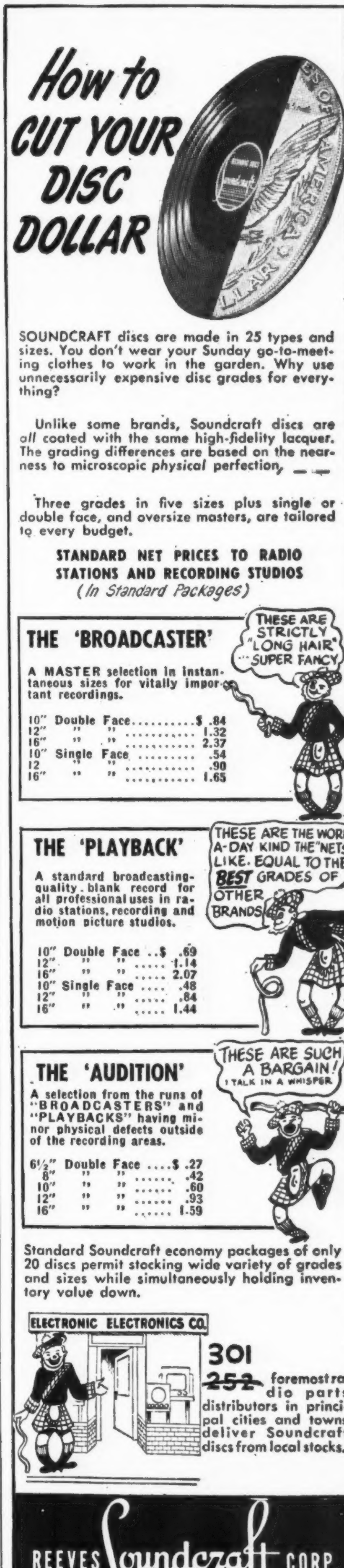
*Vice Pres., Reeves Soundcraft Corp.



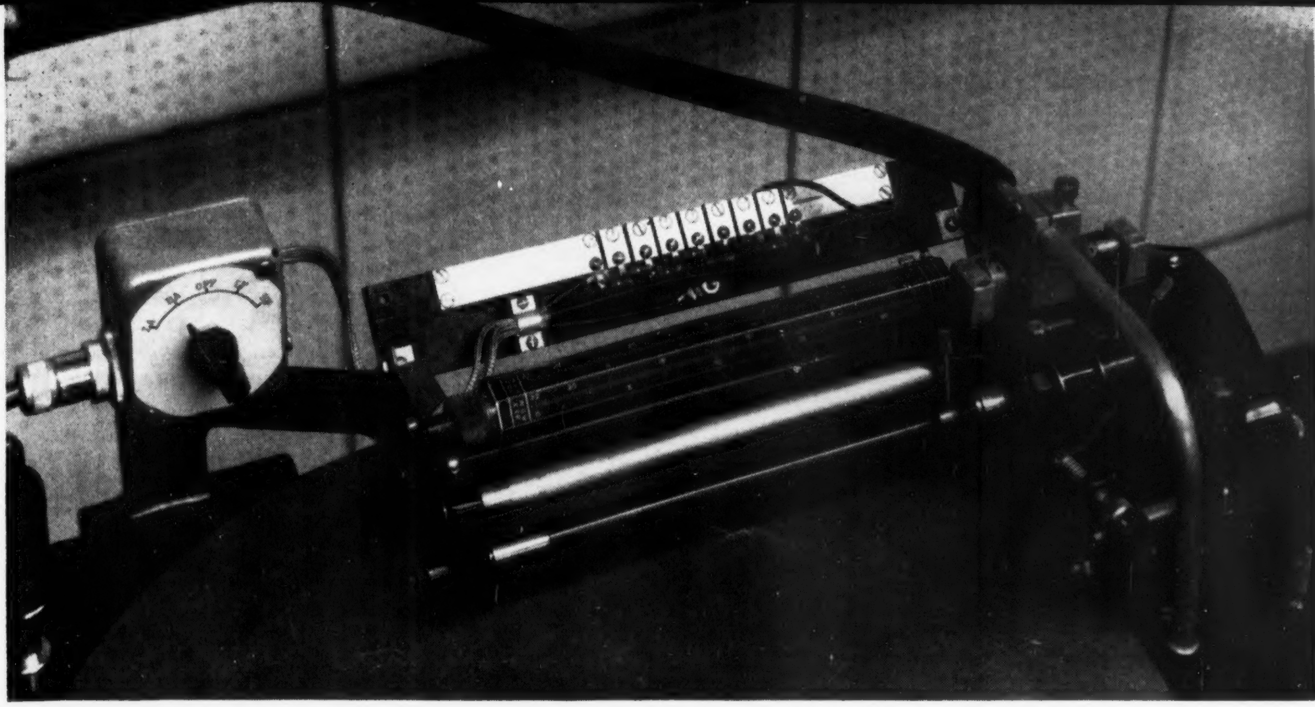


Fig. 2. Automatic radius-equalizer fabricated for Fairchild 199 recorder. Control switch is located at left of equalizer.

Disc Recording for Broadcast Stations

W. J. MAHONEY*

Technical details of a successful, high-quality studio installation.

With the recent improvements made in recording and reproducing equipment, it is now possible for medium-size broadcast and recording studios to obtain results comparable with the finest in the industry, within limitations of their acoustical studio facilities. However, there is little information available to guide the recording engineer who wishes to custom-build his installation to obtain maximum performance in each of the various services it must perform. In furtherance of this cause, the author wishes to describe the flexible recording system designed for the new studios of WSAI, Cincinnati. This paper also includes a discussion of a number of the practical problems involved in designing a complex system of recording equalizers.

The first requirement of the system was simplicity of routine operations, such as transcribing network and studio presentations for delayed broadcast or file. The second requirement was ability to record transcription masters, phonograph record masters, composite dubbings and all the various services required of a studio. In each of these special applications the proper equalization and levels must be employed. The final requirement demands a rapid method of providing substitution when equipment failures arise.

*3148 Queen City Ave., Cincinnati 11, Ohio.

Two RCA 73-B recorders and two Fairchild 199 recorders with magnetic cutters were considered adequate to handle the volume of work. Duplicate sets of amplification and control equipment were provided for each pair of machines. The only variation was the necessity to mount controls and amplifiers in one rack for the Fairchild position, due to space limitations, and the use of separate control turret and amplification rack for the RCA machines.

Program Sources

The amplification layout itself is unique. The studio control systems were complete and totally independent of the recording installation, so it was only necessary to design the recording amplification from the output of the studio system. At WSAI, which has separate control rooms, it was decided to make the recorder bus connection to shunt the channel-amplifier input at the channel side of the interlock system. Feeding from the input side of the channel was necessary because the recorder bus connection consisted of a bridging pad, which in absence of a line or 600-ohm resistive load, would not provide proper loading for a channel amplifier. The values of the series arms of the pad were 1300 ohms, the shunt arm was 500 ohms, providing a total loss of 15 db. These recording pads are permanently located in the transmission racks and provide the dual purpose of isolating the recording busses in case of short circuit, and minimizing the loading effect of paralleling many amplifier bridging-inputs.

The same type of pads was used to bridge a radio tuner system, the incoming network, and a "remote" jack on the master control bay. This remote position is used for all additional sources of program material.

Referring to the block diagram shown in *Fig. 1*, note that the input-selector switching system allows both recording and monitoring circuits to be bridged independently or in tandem across each recording bus. Following the switches are 16,000/500 bridging transformers, which provide the proper circuit for location of the standard 500-ohm "T" attenuators. The monitor and booster amplifiers fed by each attenuator are identical, with a power output of +30 dbm, and a maximum gain of 70 db.

With a normal attenuator setting, 21 db, the total losses up to the input of the monitor-booster amplifiers will be approximately 52 db. Assuming the studio is feeding a level of +8 vu, the output of the booster-monitor will be +14 vu. The monitor amplifier works

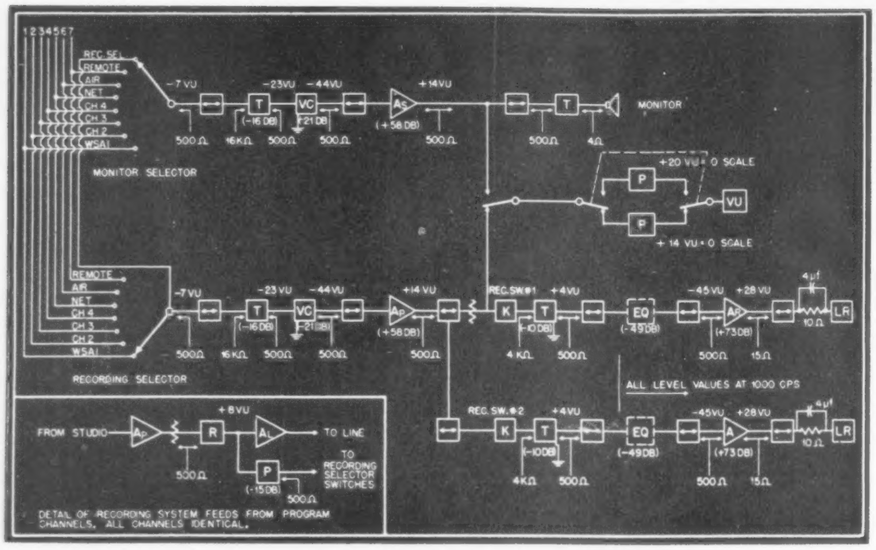


Fig. 1. Block diagram of recording installation. Program sources are fed to bridging-input pads at 8 vu.

into a speaker-matching transformer, the booster amplifier into a 500-ohm resistive load.

The volume indicator is bridged across the output of the booster, with a standard pad designed to permit the meter to read 0 vu when the booster is providing +14 dbm on steady tone. An additional pad is available on the switch, to shift the 0 vu point to +20 dbm from the booster. The additional 6 db output, obtained by advancing the master fader, is used when cutting phonograph or transcription masters. The vu meter and pads may also be transferred to the output of the monitor amplifier by a momentary-push switch. This feature, along with the ability to switch the input of the monitor amplifier independently, allows the operator to predetermine the levels to be encountered on a subsequent program from another channel without interrupting the recording in progress.

Power Amplifiers

Shunting the booster load resistor are the power amplifier input switches. The 4000/500 bridging transformers connected to these switches feed the various equalizers. The input impedance of each power amplifier is 500 ohms, with the secondary loaded by a potentiometer calculated to reflect the proper impedance. The amplifiers, Brook 12D, have a maximum gain of 80 db and a power output of 30 watts.[1]

The losses in the equalizer system are of the order of 50 db. This may seem high, but it must be remembered that some of the equalizing positions are capable of producing as much as 25 db rise at 10,000 cycles. At 200 cps, the power into the cutters is about +28 vu for instantaneous transcriptions, +34 vu or better for master recording, at 200 cycles. The equalizers have been designed so that there is no change in the level at 200 cps on any of the positions.

It will be noted that the patch field allows considerable flexibility. A tip, ring, and sleeve system is used throughout, all with double jacks in parallel except "Recording Switch #1 & #2." Substitution for input selector switch, bridging coil, fader, booster amplifier, equalizer network and power amplifier may be made by appropriate patching. Also, by use of a single patch cord from "Monitor Amplifier Output" to "Recording Switch #2," the bridging coil, #2 amplifier, and cutter may be connected to the output of the monitor-amplifier system, across the normaled loudspeaker transformer. This will allow emergency use of the two recording machines as completely independent channels. The vu meter may be transferred momentarily to the monitor channel for occasional checks of the program level.

General Design

In designing the recording system, it was felt that sufficient care should be taken to enable the installation to handle complex re-recording work and still maintain a faithful likeness to the original material. Obviously, such a system must have an accurately controlled frequency response, minimum distortion, low noise level and speed variation, and last, but fully as important as any, an intelligent operating procedure.

The distortion, noise level, and speed variation factors are inherent in the original equipment. The frequency response is reasonably subject to the control of the system designer, assuming that first-class equipment is used throughout.

In planning the equalizers, some alteration was made to the basic NAB pre-emphasis curve to provide for radius equalizing, at least on the transcriptions cut for instantaneous playback. This is a radical departure from accepted practice and, with the present degree of standard pre-emphasis (100 μ-sec), admittedly difficult to apply to all types of program material. Consequently, a compromise design was employed to permit substantially flat playback up to 5,000 cycles at the inside radius of 3¾ inches, corresponding to 15 minutes at 128 lines per inch.

Surprisingly enough, there seems to be a difference in interpretation of the NAB curve. Some authorities maintain that the present standard was set up, not only to improve the signal to noise ratio, but to compensate for the poor response of the cutters and pickups in use at that time. A new committee of standards is at work on a revision of the recording section of the NAB code, and it is hoped that a more realistic approach to the radius and pre-emphasis problem will be presented. Certainly there is no valid basis to the belief that a fixed pre-emphasis curve will overcome, in any way, the serious losses encountered with a varying radius.

In addition to the NAB equalization for transcriptions, it was desirable to have a separate pre-emphasis network for phonograph records, and a setting for flat response of the system with optional radius-equalizing.

Automatic Radius Equalizing Device

RCA has manufactured a simple automatic equalizer which has proven quite practical. It consists of several resistors, totalling 4600 ohms, soldered to a segmented rod, which in turn is wiped by a contactor attached to the cutter carriage. The contactor is wired to two different capacitance values, available through a switch, making in effect a potentiometer having capacitance in series with the variable arm. A similar device was fabricated in the shop for use on the Fairchild machines, *Fig. 2.* It consists of a dual-section RC network encompassing both radius and pre-emphasis functions and includes the automatic equalizer described above. However, it was necessary to treat the Fairchild and RCA cutters as separate problems, due to the difference in crossover frequencies of the two makes. The Fairchild, and most standard cutters, have a crossover at 500 cps. The RCA heads cross-over at slightly above 1000 cps, making a difference of about 3 db in the region from 50 to 1000 cps between the two brands.

Due to the varying impedances presented to the amplifier by the cutter head, most manufacturers provide a

[1] J. R. Edinger, High Quality Audio Amplifier with Automatic Bias Control. AUDIO ENGINEERING, June, 1947.

series resistor approximating the nominal impedance value of the head. In a 15-ohm cutter, the actual impedance varies from 2 ohms at 50 cps to 30 ohms at 10,000 cps. The constant-amplitude response of the head is determined partly by the value of the series resistance, and partly by the internal mechanical damping. The RCA heads were provided with a resistor tapped from 5 to 15 ohms, in steps of 0.5 ohms. A capacitor is used in shunt with this resistance to improve the high-frequency response of the cutter. Experiment will be necessary to determine the settings which will provide the maximum response at 10,000 cps without appreciably altering the spectrum from 2000 to 5000 cps. Values from 2 to 4 μf will usually be necessary in a 15-ohm circuit. As the compensator supplied with the Fairchild recorder has no easy adjustment, it is difficult to use this method of altering frequency response.

Playback Standard

Unless the designer has an instrument such as the FM Calibrator[2] or a reliable method of optically measuring the stylus tip, it will be necessary to set up a playback device capable of acting as an accurate standard for frequency measurements. The light-pattern method[3] of measuring a severely pre-emphasized cutter response is far too tedious and inaccurate, although valuable for constant-velocity measurements. By means of careful light-calibration of the Columbia test record YTNY-170 it was determined that a Pickering pickup was reliable within 1 db at all frequencies covered on the test record. Thereafter, the Columbia record was used as a control to ascertain that the characteristics of the pickup were not changing. It is also advisable to record your own light pattern test-record as a double check. The next step in setting up the standard was careful plotting of the NAB playback equalizer. This was accomplished by use of a variable frequency oscillator and a gain-set capable of inserting sufficient loss to equal the voltage from the pickup. By adding this source in series with the inductance of the pickup and measuring the equalizer through the playback amplifier, an accurate reading of the network may be made[4]. The response of this circuit was found to be within 0.5 db of the complement of the NAB pre-emphasis curve. While set up for this measurement, it will be wise also to check the flat-response position of the equalizer, because some networks will be found to start the bass boost above 1000 cps, thus throwing off all readings above this frequency by 1 or 2 db. Additional standard equipment should consist of a good variable-frequency oscillator and a gain-set capable of presenting proper transmission characteristics to the amplifying system. The gain-set output meter should be accurate over a range of 30 db.

Before beginning the design procedure, it will save much confusion if several styli are tested and set aside for measuring purposes. Even a newly sharpened stylus, with slightly dull burnishing facets, may give a reduced response from 8 to 10 db at 10,000 cps. A recommended procedure is to cut bands of 8 kc, 9 kc, and 10 kc of exactly the same light pattern width as 1000 cps. With a calibrated pickup, these frequencies should not show any deviation when the playback filter is set for flat response. Obviously, these styli must also be acceptable for quietness of cut, but almost invariably the satisfactory points will possess both attributes. A short-shank stylus is desirable, as the long-shank type possesses an undesirable mass resonance at 8000 cps. We do not wish to enter into the discussion of 70° versus 87° styli, but it should be pointed out that there is a difference in the ability of a playback stylus to reproduce high frequencies from the different cutting angles. We have found the 70° angle to have about 2.5 db higher response at 10,000 cps when played with a .0025" radius reproducing point, with a slight increase in distortion between 1000 and 5000 cps. Of course, when recording phonograph record masters, the wide angle stylus is mandatory.

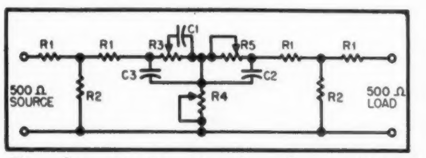


Fig. 3. Basic circuit of combined NAB and radius-equalizing networks. (Values explained in text.)

Heating Effect

When the recording heads are subject to signal, there will be some increase in efficiency due to the heating of the damping material. The resultant increased output is therefore considerably more evident in the damped, or constant-amplitude, portion of the curve. Because of this, the reference tone may be found to be as much as 2 db higher when repeated somewhere in the middle of a frequency run. The RCA MI-11850-C head is equipped with a built-in heater unit and thermostatic control to minimize this heating effect on program material; however, this is not sufficient protection to warrant disregarding the effects of steady tone during measurement. For this reason it was found necessary to resort to alternating two-second tone bursts with fifteen seconds of program at normal level when working within the 4000 to 10,000 cps region of the equalized curve. In this spectrum the amplifiers will be delivering 8 to 16 db higher power levels than the unequalized portion of the curve. For additional protection, the tone input to the equalizer system should be at least 6 db below the meter peaks of the program material. When working in the 50 to 2000-cps region, such a technique of alternation is not essential if the tone is kept considerably lower than normal program peaks. Incidentally, comparable troubles may be encountered on non-temperature-controlled heads due to ambient temperature changes. The author had a disagreeable experience when trying to duplicate a previous day's work on the following morning with an ambient temperature change of only 10 degrees. To avoid this trouble, it is good practice to introduce program at normal level for at least thirty minutes before attempting any measurement.

As a last word of caution before proceeding to the actual equalizer design, it should be impressed upon the reader that *all* measurements from 4000 to 10,000 cps at 33 1/3 rpm, *must* be made at the outer edge of a 16" disc, unless actually working upon the problem of radius losses. The 10,000-cps playback response will drop 1.5 db within one inch of the edge, and will drop 9 db at three inches with .0025" radius stylus.

RADIUS (inches)	7.5	6.75	6	5.25	4.5	4	3.5
1000~	0	-.2	-.5	-.6	-.7	-1.3	-1.5
5000~	0	-.7	-1.9	-3	-4.6	-7.2	-10.2
7500~	0	-2.2	-4.5	-7.7	-11.7	-17.7	-22
10000~	0	-2	-6	-12	-20	-21	-25.2

Fig. 4. Typical radius losses encountered on playback with RCA Vertical-lateral Pickup .0025" radius stylus.

Also, because of these radius losses, checks involving comparative portions of the high-frequency spectrum should be made at as nearly the same radius as possible. In the further interests of uniformity, the experimenter should use the best brand of discs, and preferably from the same package, in order to minimize the chance of error due to non-uniform coating consistency. The "softer" types of coatings will usually produce a slightly reduced

[2] R. A. Schlegel, FM Calibrator for Disc Recording Heads. AUDIO ENGINEERING, June 1947.
[3] C. J. Lebel, Light Pattern Calibration Chart. *Communications*, April 1940. Extended Experimental Study of Optical Pattern *Communications*, December, 1940.
[4] C. G. McProud, Elements of Residence Radio Systems. AUDIO ENGINEERING, November, 1948.

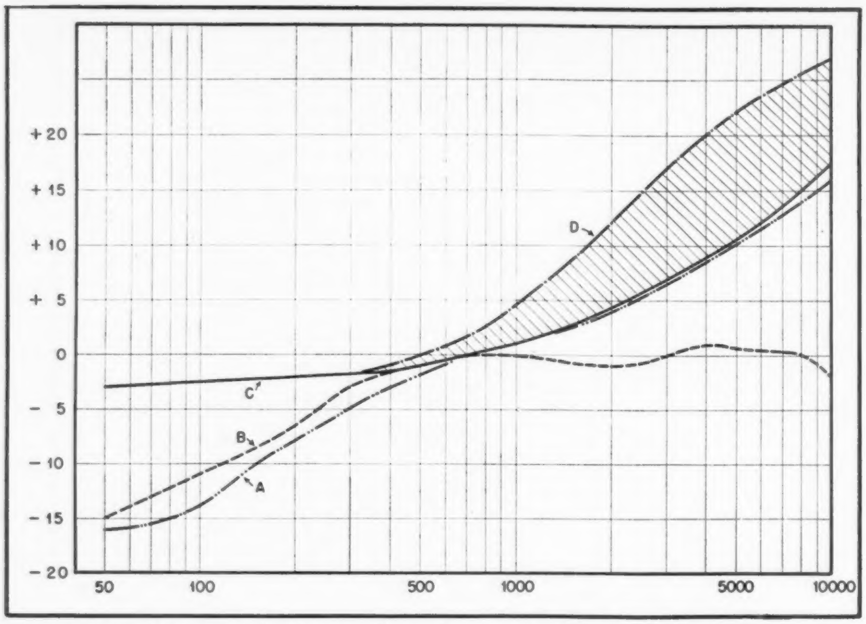


Fig. 5. A—NAB standard recording characteristic. B—Typical response of cutter with 500 cycle crossover. C—Response of NAB position on equalizer shown in Fig. 7 with radius-equalizer at minimum or off. Shaded area-Action of automatic-equalizer, continuously variable to the maximum shown by line D.

high-frequency playback response, approximately 1.5 db at 10,000 cps, probably due to the elasticity of the coating under the tremendous accelerations of the pickup stylus.

Network Design

In proceeding with the network design, the author desired to insert the experimental equalizer in the physical portion of the circuit where it would finally be used. In this way it is possible to compensate directly for all high-frequency losses due to long cable runs, amplifier deficiencies, and so on. The test signal was introduced into the front of the entire recording circuit at the program selector switch, and readings were taken from the output of the recording amplifier. Inasmuch as the selector input switch feeds a bridging coil, the oscillator or gain-set must work into a 500- or 600-ohm resistive load. For accurate measurement purposes the recording amplifier must feed a resistive load instead of the inductive load of the cutter.

The experimental form of the equalizer, with the exception of the fixed isolation pads at the input and output of the equalizer, consisted of variable carbon potentiometers and an assortment of small tubular paper capacitors ranging from .001 to .01 μf. All values of capacitance within this range should be covered in steps of .001 μf. Also, several capacitors from .01 to 0.5 μf will be necessary. Three or four of each value should be available, with actual measured values marked on the outside. Considerable later confusion will be avoided by this precaution.

To isolate the inductive and capacitive effects of the transformer windings adjacent to the equalizer, T-pads should be used at both ends of the network. An attenuation of 9 db is sufficient to provide complete isolation, but it is possible to use only 6 db attenuation if the over-all network losses become too great.

Two methods may be used to provide the proper working load for the equalizer and T-pads. If the input transformer to the recording amplifier must be operated without secondary loading, due to inclusion within feedback loops, for example, then the primary winding must be loaded with the proper value of resistance (in this case, 500 ohms). However, if the secondary can be loaded, then the step-up ratio of the transformer should be figured and the proper resistance shunted across it to reflect 500 ohms to the primary.

It may be convenient to utilize the secondary loading resistance as a potentiometer to control the gain of the amplifier. Loading the secondary aids in reducing distortion, and is preferred whenever it can be done conveniently.

The layout of the equalizer is shown in *Fig. 3*. In its experimental form R_3, R_4, and R_5 are potentiometers; R_1 and R_2 are the series and shunt arms of the isolation pads. Values of 2000 ohms for R_4 and 20,000 ohms for R_5 will allow sufficient range for test purposes. R_3, substituting for the radius equalizer, should be 5000 ohms.

The radius losses should now be charted. Without any pre-emphasis equalization, record bands of 1000, 5000, 7000 and 10,000 cps at radii of 7, 5, 6¾, 6, 5¼, 4½, 4 and 3½ inches. Tabulate the playback readings of these test cuts, as in *Fig. 4*. The desired response of the radius equalizer will now become apparent. Usually the losses at the smaller radii will be too great for any practical amount of compensation to overcome entirely, especially when added to an already pre-emphasized response. It is satisfactory to employ sufficient equalization to maintain flat playback to about 5000 cps at these inner radii.

Computation of NAB Equalizer

It is now possible to compute the equalization necessary to provide the equivalent of the NAB recording curve. The responses of the cutter itself should be taken by plotting the playback readings from the calibrated pickup. Test frequencies should be 50, 100, 200, 500, 800, 1000, 2000, 4000, 8000, 10,000 cps with the highest frequency at the outer edge of the disc. By adding the irregularities of the cutter algebraically to the NAB standard curve, the desired response of the equalizer will be shown. For example, if the cutter is -3 db at 10,000 cps where the NAB standard calls for +16 db, the desired response of the equalizer should be +19 db. Thus the *reproducing complement* of the NAB pre-emphasis curve will be met, and the system will produce a flat playback response from any standard reproducing system. In these computations the designer should be careful to use 800 cps as the reference point. Of course, a cutter that departs from flat response at the high end of the spectrum by more than 4 or 5 db cannot be fully compensated, and it is also impractical to attempt to neutralize large peaks or valleys within the spectrum by this method.

The most crucial problem will be the treatment of the region from 100 to 1,000 cps. Any departure from flat playback in this spectrum, which includes most music and speech fundamentals, will be serious when the system is called upon for complex dubbing work. If the ideal curve of a recording head with standard 500-cps crossover is plotted against the NAB curve, *Fig. 5*, it will be seen that the spectrum from 50 to 500 cps will be higher than the NAB curve by about 2 db. To correct this, the equalizer must start to rise as low as 300 cps and reach +3 db at 1000 cps.

However, with any combination of R_4 and R_5 that did not provide too much loss for the system to handle, such a curve became asymtotic at about 5000 cps when the proper capacitance C_2 was employed to provide the correct slope from 3000 to 4000 cps. In order to keep the response rising from 5000 to 10,000 cps, it proved necessary to add

an auxiliary capacitance, C_3, across the entire radius-equalizer resistance. Obviously, any capacitance added here detracts from the effectiveness of the automatic radius-equalizer unless the shunt resistance R_4 is lowered. This, in turn increases the loss of the network, and a constant check of the output level from the recording amplifier must be made while working with trial values to ascertain that sufficient power is still available to drive the cutter. The radius-equalizer also must be observed periodically to determine that the curve is capable of being increased by the amount necessary to add full equalization up to 5000 cps at the inner radius.

The final values of the components in the equalizing system, as set up on breadboard, should be capable of providing a curve to fulfill the requirements of both NAB pre-emphasis and radius losses. After these values have been ascertained, the size of the series resistors between the segments of the automatic radius-equalizer may be determined. In order that the over-all gain of the system should not change by more than 1 db when the sliding contactor momentarily shorts two segments, the value of each resistor should be kept small. The device built at WSAI was divided into nine segments, providing a change of response every 1½ minutes. The values of resistance necessary were found by measuring the settings of potentiometer R_3 when adjusted to give the desired response at the various radii. In regards to radius losses, it was found that the combination vertical-lateral pickups used in the regular studio turntables were less efficient than the Pickering Pickups used in the re-recording setup. Therefore, use was made of the switch supplied with the automatic radius-equalizer to change capacitance values, in order to provide a less steep curve for transcriptions made expressly for copying. Additional positions were provided on switch to make the maximum radius-equalizer responses available as fixed curves.

Network For RCA Cutters

The network designed for use on RCA recording heads will differ from the above only in the detail of treatment to the spectrum from 100 to 2,000 cps. With the high crossover frequency, 1000 cps, it will be noted that the constant-amplitude portion of the curve already fits the NAB Standard *before* any pre-emphasis is added. Inasmuch as it is impossible to make a rising curve of 4 db per octave with a total of 15 db from 1000 to 10,000 cps and which has *no* effect on the region from 1000 cps down, the real problem is to minimize any change in the lower spectrum. Fortunately, most RCA cutters have a 2 db rise at 2000 cps. By taking advantage of this existing hump, it is only necessary to design the network with a slow beginning rise and a rapid increase at the high frequencies. This can easily be done by adding auxiliary capacitance across the radius equalizer resistance R_3.

Listening tests indicated that a slightly less steep pre-emphasis curve seems to be in general use today for phonograph records. A 75-μsec network, with a total rise of 10 db at 10,000 cps sounds quite acceptable. A value of C_2/R_5 was chosen which provided such a curve. The value of R_4 was so adjusted as to hold the signal level of the unequalized portion of the frequency spectrum at the same level as the NAB position. Inasmuch as it is impractical to use radius equalization with phonograph records, the automatic equalizer is turned off in this service. However, it is still available if desired.

A flat position of the equalizer network was also provided for test and other purposes. Without pre-emphasis, it is practical to include a greater degree of radius equalization at the inner diameters. The section C_2-R_5 is removed and the loss increased in the rear pad, so that there is again no increase in over-all level when this position is used. It is practical to find the necessary pad value by substituting a standard variable attenuator and reading the loss directly from the dial.

Due to the necessity of keeping the automatic equalizer common to all positions of the selector switch, the auxiliary capacitor C_3, used on the NAB position, became shunt to ground through 800 ohms when the switch was moved to another position. The resultant high-frequency attenuation could be counteracted on the PHONOGRAPH position by increasing the value of C_2. However, on the FLAT position the presence of C_3 caused 4 db attenuation at 10,000 cps. To correct this, it became necessary to shunt the series arms of the output pad with a .04-μf capacitor. *Figure 6* is a schematic of the entire equalizer and *Fig. 7* shows the control panel.

Operating Techniques

The operating staff was introduced to several techniques which enable the studio to produce competent work. For instance, in handling quantities of direct-copy material, the original record is made with the cutter carriage in a direction of travel opposite to that of the finished copy. This, in conjunction with the action of the automatic equalizer, produces an amazingly faith-

(*Continued on page 45*)

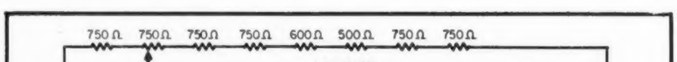
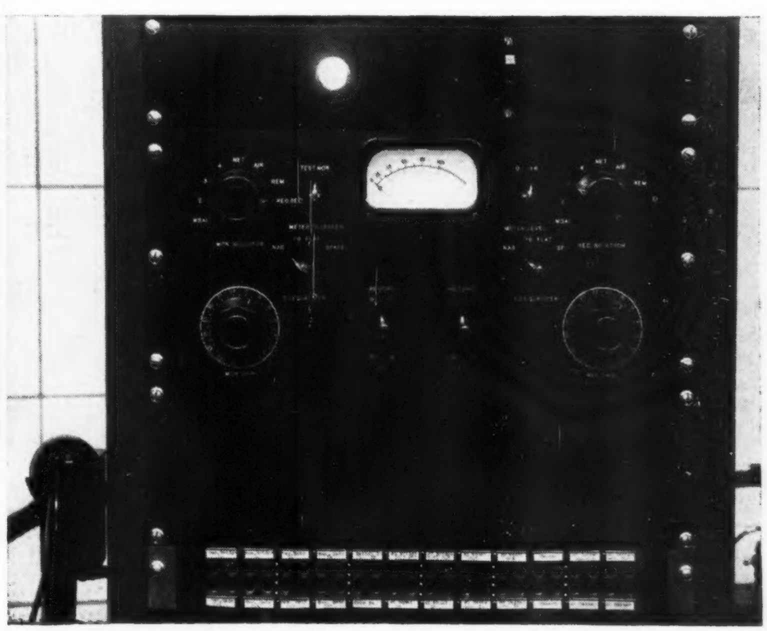


Fig. 6. Complete equalization system for 500 cps crossover cutter, showing NAB phonograph and flat equalizers, and the two-position radius equalizer available for each curve.

Fig. 7. Control panel and patch field of recording bay. All bays are identical.



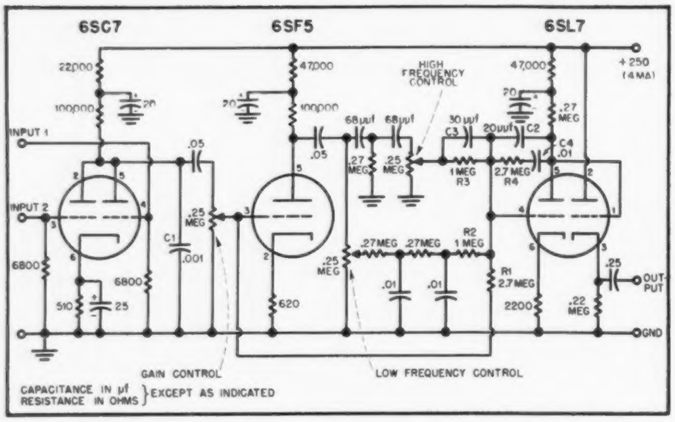


Fig. 1. Schematic of versatile pre-amplifier with two phono inputs, cathode-follower output.

A Continuously Variable Equalizing Pre-amplifier

DAVID C. BOMBERGER*

THE reproduction of phonograph records has long been complicated by a lack of uniformity in the recording characteristics. While broadcast transcriptions have been fairly consistently made with a standard low end roll-off and high end pre-emphasis, the recordings available to the general public have not been so standardized. As a result, it is difficult to choose a frequency characteristic for reproduction which will produce satisfactory results for all commercial pressings. The advent of microgroove recordings, with identical characteristics as published, promises hope for the future, but at the moment it further complicates the problem by adding yet another characteristic to the list.

One evident solution to the problem is an adjustable equalizer which permits the reproducing characteristic to complement that of the recording, whatever it may be. This is practicable only when all the required characteristics are known. Lacking knowledge of the exact requirements, the approach may be a continuously variable equalizer which will approximate all possible recording characteristics. Then an adjustment may be made by ear. While this method does lack exactness, it can lead to aural satisfaction which, after all, is its purpose.

Equalization

Equalization is normally achieved by what might be considered a "multiplication" process. By this is meant that the gain-frequency characteristic (in the form of input-output ratio) is multiplied by some function of frequency. Any reasonable function of frequency is obtainable with R, L and C networks, but a continuously variable function of frequency is another matter.

An alternative method of equalization might be termed "additive." That is, a rise in gain at some frequency, for example, is accomplished by the addition of extra voltage at that frequency. To produce an equalizing characteristic, a function of frequency must be applied to the amplitude characteristic of the added voltage, and the addition will thus have phase shift which varies with frequency. The design process, therefore, involves vector addition instead of the vector multiplication of the ordinary design. While this process is somewhat troublesome, the realization of the design in circuit form is readily accomplished, since the variable control element is a potentiometer.

The continuously variable equalizing pre-amplifier presented here utilizes the additive method, with three transmission channels. One channel has a characteristic which is essentially flat at frequencies below 1000 cps, and falls off at the rate of 12 db per octave at higher frequencies. This is the basic channel, to which voltages from the auxiliary channels are added. The second channel has, at very low frequencies, 40 db more gain than the basic channel, but its gain falls at the rate of 12 db per octave above about 50 cps. The third channel gain rises at the rate of 12 db per octave up to 15,000 cps; above this frequency its gain is also 40 db more than the basic channel. Outputs from the three channels are added in a single tube feedback summing amplifier; potentiometers which add flat loss in the auxiliary channels permit control of the resultant transmission characteristic.

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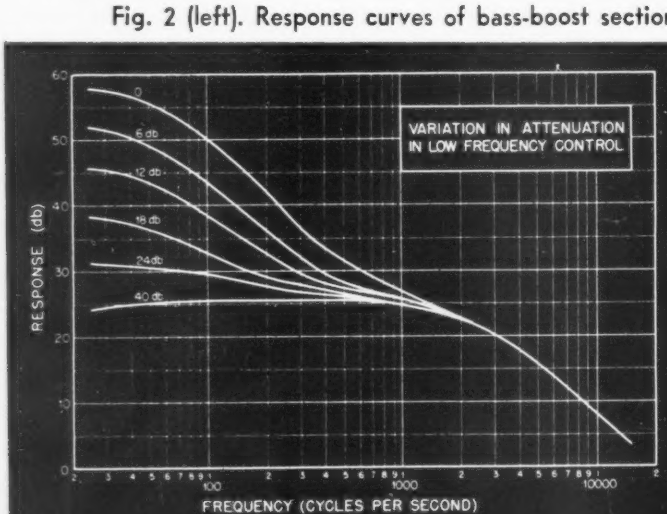
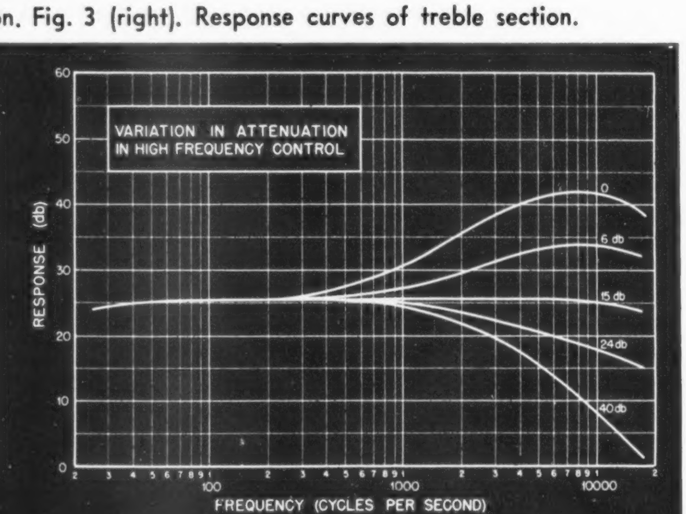
Fig. 2 (left). Response curves of bass-boost section. Fig. 3 (right). Response curves of treble section.





The pre-amplifier schematic is shown in *Fig. 1*. In this particular embodiment, as used in the author's home, the input is a double triode, to accept signals from two variable reluctance phonograph pick-ups simultaneously. After amplification in this tube, the signals are fed to a potentiometer for gain control, and then split into two channels. One of these, the basic channel, goes directly to the summing amplifier. The other, which is further amplified by a triode stage, is subdivided into the two auxiliary channels by low-pass and high-pass *RC* networks. These networks produce frequency functions suitable for additive equalization.

The feedback amplifier in which the three channels are added has a capacitor, C_2, across its feedback resistor, R_4. This, and the capacitor C_1 shunting the plate circuit of the input tube, cause the 12 db per octave high-frequency cut-off of the basic channel. The input resistor, R_3, for the high-frequency auxiliary channel has a capacitor, C_3, in shunt for partial compensation of the effect of these elements on the high-frequency auxiliary voltage.

The summing amplifier output is direct-coupled to the grid of a cathode follower, which permits the use of the pre-amplifier at a moderately remote location. The maximum output of the pre-amplifier is not limited by the output stage, which is capable of swinging almost 200 volts peak-to-peak. Overload occurs first at the grid of the auxiliary channel amplifier, which can swing only about one volt peak-to-peak. Since the mid-frequency gain from this grid is unity, the maximum output is also one volt peak-to-peak. This is comparable to the output from a crystal pick-up, insofar as voltage is concerned. Because of the low output impedance of the cathode follower, the output power level of the pre-amplifier is considerably above that of a crystal.

The measured performance of the pre-amplifier is shown in *Figs. 2, 3* and *4*. *Fig. 2* shows the effect of varying the attenuation in the low-frequency channel, with large attenuation in the high-frequency channel. In *Fig. 3* these conditions are reversed. It is seen that the two auxiliary channels overlap in the region of 500 to 1000 cps. The effect of this overlap is seen in the composite characteristics of *Fig. 4*; it is less than 2 db for attenuation greater than 15 db in the auxiliary channels. Smaller attenuation will be used only to compensate unusually poor recordings, or to create special sound effects.

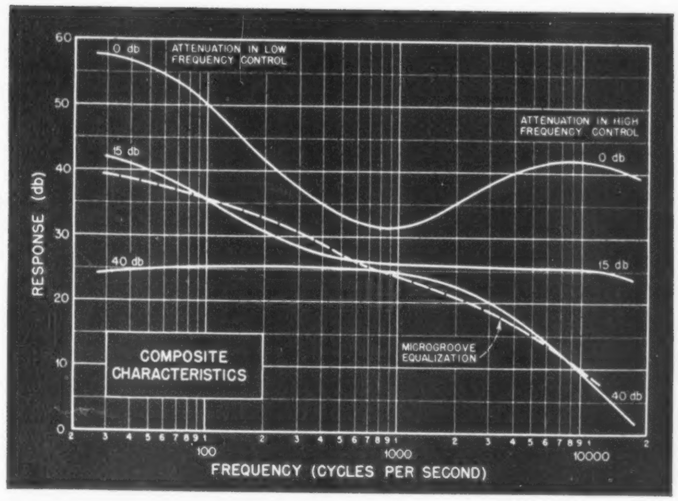


Fig. 4. Limits of available response curves.

Only three characteristics are shown in *Fig. 4*. Two of these represent extremes, with either no attenuation, or large attenuation, in both auxiliary channels. Between these extremes is a large number of possible characteristics, of which only one is illustrated.

The required equalization for microgroove recordings is also shown in *Fig. 4*. This characteristic may be achieved within a variation of ±2.5 db with attenuation settings of 15 db in the low-frequency channel and 40 db in the high-frequency channel.

The method of equalization presented here is not new. Circuits utilizing the addition of several channels have been described before, but meagre performance data were given. The data shown here are the results of measurements made after the circuit was tailored to produce what were considered to be desirable characteristics. Experience with the pre-amplifier has given convincing evidence that this degree of flexibility is profitable.

N.A.B. Convention

THE 27TH ANNUAL CONVENTION of the National Association of Broadcasters will be held in Chicago from April 6 to 13, with the Stevens Hotel serving as headquarters. Of particular interest is the Broadcast Engineering Conference which occupies the first four days of the convention.

On April 6, the opening day, the afternoon will be given over to a tour of the Hallicrafters plant, at which ladies will be welcome. The papers to be given on Thursday morning cover television problems, from the selection of a transmitter site up to operation of the image Orthicon camera, and including the description of a 2,000-mc relay link.

The principal audio papers are scheduled for Thursday afternoon, and are:

- **AM, FM, and TV Audio Measurements**, Frank H. McIntosh, Consulting Radio Engineer, Washington, D. C.
- **The NAB Recording & Reproducing Standards for Disc, and Magnetic Recording**, Robert M. Morris, Radio Facilities Engineer, American Broadcasting Co., New York.
- **Magnetic Tape Recording and Reproducing**, Dr. S. J. Begun, Vice-President in charge of Engineering, Brush Development Co., Cleveland, Ohio.
- **Properties of Magnetic Tape and Their Relation to Magnetic Recording**, Reynolds Marchant, Development Engineer, Magnetic Tape Equipment, Minnesota Mining & Manufacturing Co., St. Paul, Minn.
- **A New Portable Audio Amplifier for AM-FM-TV**, William W. Dean, Audio Engineer, Broadacst Engineering Section, General Electric Company, Syracuse, N. Y.

The official opening of the exhibits is at 5:00 p.m. in the Exposition Hall of the Stevens Hotel, with ladies again being invited. Smaller equipment will also be on exhibition on the fifth and sixth floors of the hotel during the entire convention week.

The Friday morning session involves transmitters and problems associated with them, except for one paper to be given at 11:15 a.m. on Automatic Selection of Broadcast Program Circuits by John A. Green of Collins Radio Co., Cedar Rapids, Iowa, to be followed by a demonstration to be given by Mr. Robert D. Essig of the same company. The afternoon session again deals with television, while the Saturday morning

[*Continued on page 37*]

Compact 6AS7G Amplifier For Residence Audio Systems

C. G. McPROUD*

Part II—Further constructional details on two-unit amplifier of unique design.

In the first section of this series, a new type of amplifier was described. This amplifier employs a novel arrangement for furnishing plate power for the output stage as well as the d-c filament power for all the other stages. While the design was discussed thoroughly, exact component values were omitted pending the results of the measurements of distortion and overall performance. These measurements have justified preliminary tests, and the components are listed herein.

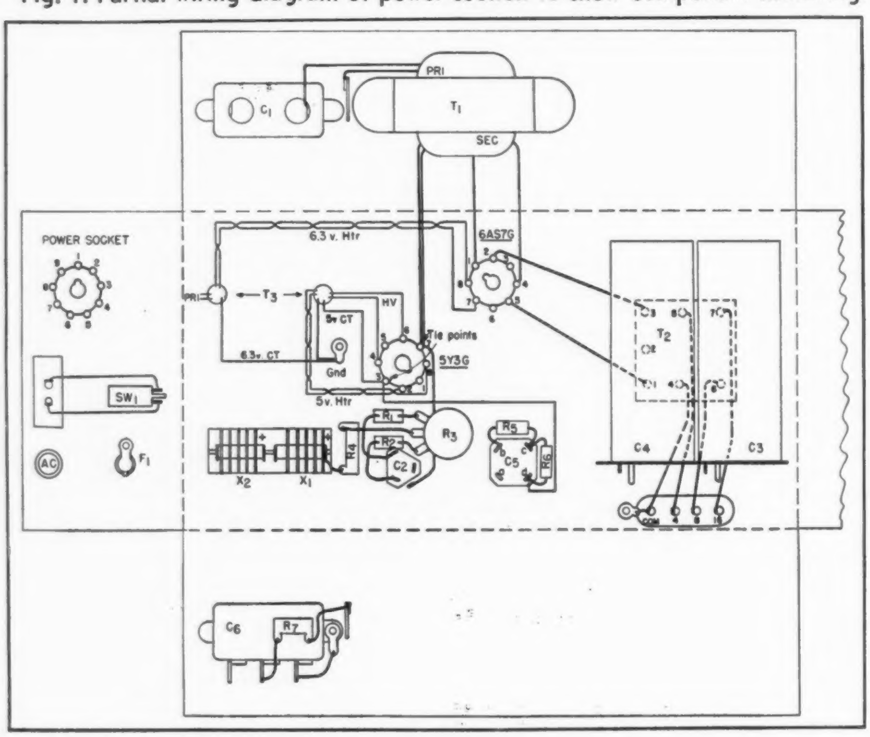
The results of the performance tests are gratifying. Power output at one per cent harmonic distortion is 6.5 watts at 400 cps, 6.2 watts at 20 cps, and 6.35 watts at 20,000 cps. Eight db of feedback is employed, extending from the secondary of the output transformer to the cathode of V_{2b}. The output impedance on the 16-ohm tap is 1.85 ohms, which gives excellent damping. The frequency response curves were shown in Part I, with the tapped tone switches in various positions. Hum and noise measures −42 dbm, which is not exceptionally low, but which is within the range of good quality amplifiers.

With the volume control at maximum—which is the operating point for the greatest room volume normally desired—a two-volt input signal is required at the two radio input jacks for a two-watt output. This does not leave much leeway, but will suffice for most tuners. The phonograph preamplifier supplies the additional gain to bring the output of magnetic pickups to the equal of the radio inputs.

The original design provided for a roll-off in the LP phonograph position so that these records would reproduce normally with the tone controls in the positions for flat reponse. However, this does not provide sufficient gain, so it is considered more desirable to eliminate R_{12} and C_5. Values for these components are given in the list of parts, but it is recommended that the arm of SW_{1b} be connected directly to the LP position contact of SW_{1c}, as shown in the schematic, *Fig. 4*. It should be mentioned at this point that the elimination of this equalization arrangement does not prevent the user from reproducing LP records at the correct tone control settings, because position 4 of the high-frequency tone control gives the correct roll-off for these records.

*Managing Editor, Audio Engineering

Fig. 1. Partial wiring diagram of power section to show component mounting.



List of Materials

C_1, C_2, C_{14},	0.05 μf, 400 v, paper
C_3	.003 μf, mica
C_4, C_8	.002 μf, mica
C_5*, C_6	250 $\mu\mu$f, Centralab Hi-Kaps
C_7, C_9, C_{10}, C_{11}	750 $\mu\mu$f, Centralab Hi-Kaps
C_{12}	.006 $\mu\mu$f, mica
C_{13}	0.1 μf, 400 v, paper
$C_{15}a$, b, c	15-15-10/450 electrolytic
R_1,	10,000 (all values ½-watt unless otherwise specified)
R_2, R_5, R_{29}	2200
R_3, R_6	0.12 meg, 1-watt
R_4	1.0 meg
R_7	0.1 meg
R_8	47,000, 1-watt
R_9, R_{10}, R_{11}	0.5-meg potentiometer
R_{12}*, R_{17}	0.12 meg
R_{13}, R_{14}, R_{15}, R_{16}, R_{22}	0.27 meg
R_{18}	82,000
R_{19}	68,000
R_{20}	33,000
R_{21}	0.39 meg
R_{23}, R_{24}	0.18 meg
R_{25}	56,000
R_{26}	1000
R_{27}, R_{32}	0.1 meg, 1-watt
R_{28}	special volume control (see text)
R_{30}	0.56 meg
R_{31}	2700
R_{33}	82,000, 1-watt
R_{34}	22,000, 1-watt
SW_1	Mallory 3136J
SW_2	Mallory 3115J, modified (see text)
SW_3	Mallory 3115J
V_1, V_2	12SL7
V_3	12SJ7

*These components should be omitted—see text.

The parts not specifically described for the power section are as follows:

C_1	0.5 μf, 600 v, oil filled, bathtub type
C_2	40-40-40/150, electrolytic, with insulating tube
C_3, C_4	125 μf, 350 v, electrolytic, with insulating tube
C_5	40-30-20-10/450, electrolytic
C_6	0.1-0.1, 600 v, oil filled, bathtub type
R_1, R_2	600, 5-watt
R_3	500-ohm wire-wound potentiometer
R_4*	15,000, 10-watt
R_5, R_6	6800, 2-watt
R_7	5, 5-watt
SW_1	SPST toggle switch

T_1 push-pull input, special channel mounting, no d.c. in primary; Freed 17290
T_2 4000 ohms plate to plate, 4-8-16-ohm secondary; Freed 15929
T_3 325-0-325 v at 50 ma; 5v at 2a; 6.3v at 2.5 a. Freed F-413
X_1, X_2 200 ma selenium rectifier
*Shown on diagram as 7500 ohms. This value should be adjusted to provide 36 volts across filament string.

The performance of any amplifier depends to a large degree upon the quality of the components used. At the time this amplifier was designed, it was desired to use high-quality transformers throughout, and to make the power section as compact as possible the input transformer had to be mounted underneath the chassis. This ruled against a cased type, and few manufacturers list high-quality transformers in open-frame mounting. The unit employed was designed to work between a single 12SJ7, triode connected, with no d.c. in the primary, and to obtain adequate driving voltage for the grids of the 6AS7G it was required that the step-up ratio should be fairly high. To get this performance and retain a wide frequency range, the transformer is wound in two sections, both placed on the center-leg of a conventional E-I core. Measured frequency response of the transformer itself indicates a droop of 1 db at 30 cps and at 30,000 cps. The output transformer, also special, shows a droop of 1 db at 17 cps and at 120,000 cps when operated without feedback, and drooping 1 db at 15 cps and 62,000 cps when operated with feedback. Similar output transformers are available in the standard Freed line under number F-1951 with output impedances of 1.2 to 30 ohms, and under number F-1950 for impedances from 50 to 500 ohms.

The following table indicates transformers of high quality which are generally obtainable from jobber stocks and which should perform satisfactorily, since their characteristics are similar to those used in the original amplifier.

Construction Hints

There are a number of suggestions which may be of interest in the construction of these two units. Referring first to the power section, it will be noted that the two filter capacitors, C_3 and C_4, are mounted on a bracket under the ouput transformer. Therefore, they are necessarily installed after the wiring to the output transformer is completed. The balancing potentiometer, R_3, is also mounted on a bracket so that its shaft may be adjusted through a hole on the chassis between the two capacitors mounted on top. The selenium rectifiers are mounted on a 6-32 threaded rod which passes through two strips of Bakelite attached to bent-up angles on the chassis. Two saw-cuts are made ¾ in. apart and extending for two inches along the chassis. At the center of these two cuts, another cut is made between them. This frees two "flaps" which may be bent up to mount the Bakelite strips. The cover is bent up from perforated metal to prevent accidental contact with hands or tools.

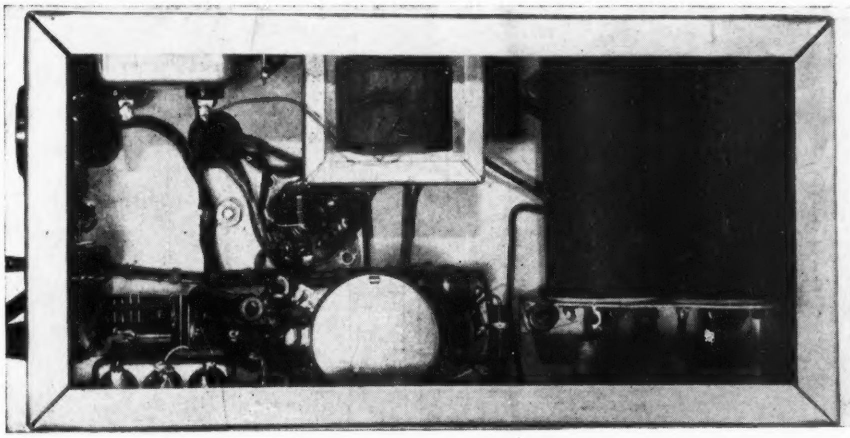


Fig. 2. Underside of power section chassis.

The value for the resistor R_4 was shown as 7500 ohms. This gives somewhat too much current through the filament string, and it is found that 15,000 ohms is more suitable, since the voltage across the filaments should be 36 volts. R_7 has been added to reduce the peak current through the selenium rectifiers, and 5 ohms is a suitable value. This resistor is connected between SW_1 and C_3.

The push-pull input transformer is mounted directly below the 6AS7G socket, using leads as short as possible. This will normally require that the leads be connected before the transformer is bolted in place, and since the leads are likely to be of relatively light wire, care should be exercised in this operation. The two bathtub capacitors are mounted on opposite sides of the chassis, with tie-points installed adjacent to them. One is needed to make the connection to the plate end of the primary, and the other serves to hold R_7.

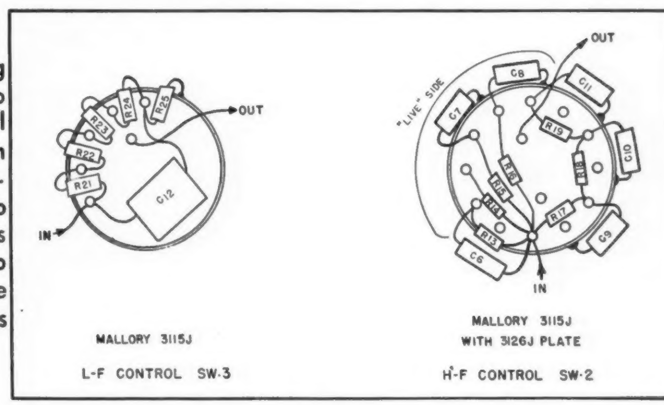


Fig. 3. Wiring diagram of two tone-control switches. Switch plate on H-F control is changed to furnish tie points on five-step frame. Only one contact arm is used.

With some transformers it is probable that there will be a tendency to oscillate at some super-audible frequency. This may require some experimentation, but it is suggested that a small capacitor across each half of the primary of the output transformer, or possibly across the secondary of the input transformer, will suffice to eliminate this trouble. Probable values

TABLE I

	T_1	T_2	T_3
Audio Development Co.	214H* 215C*	314C 315F	515C (large)
Chicago Transformer Div.	— —	BO-6	PCC-70
Stancor	A-4750	A-3800	P-4078
Thordarson	T20A22	T22S70	T22R02
UTC	LS-21* CG-132*	LS-57[1] LS-55[2] CG-16	R-54

*Insufficient space under chassis to mount these models.
[1]Voice coil secondary only. [2]Voice coil and line secondary.

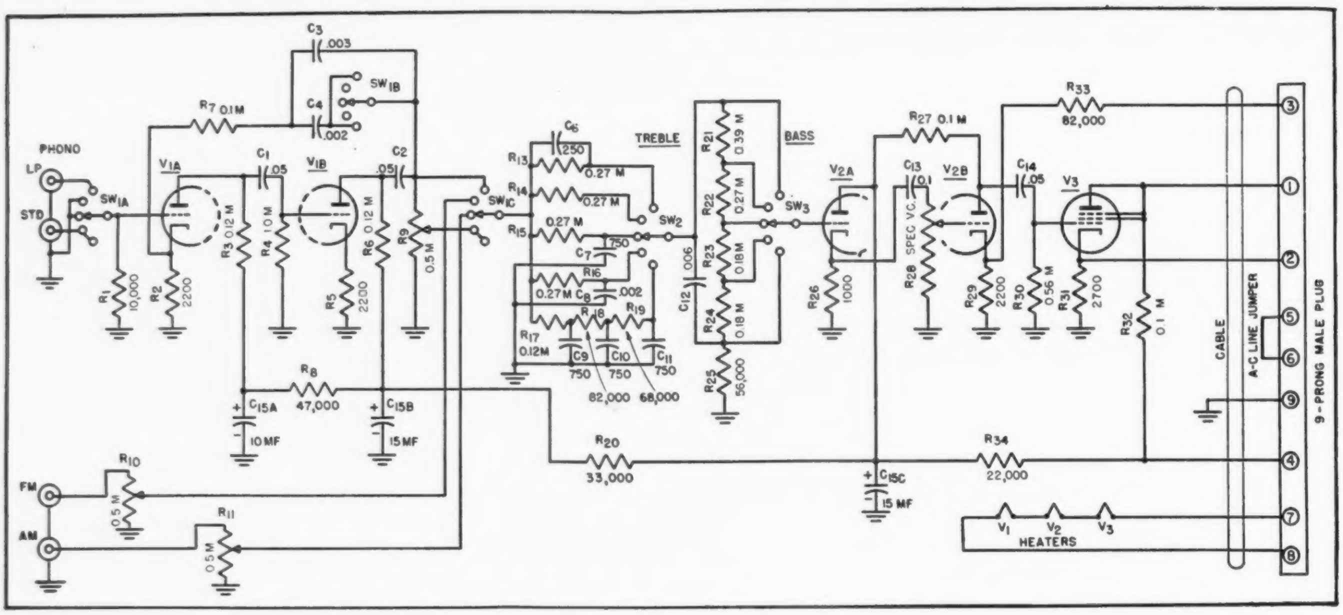


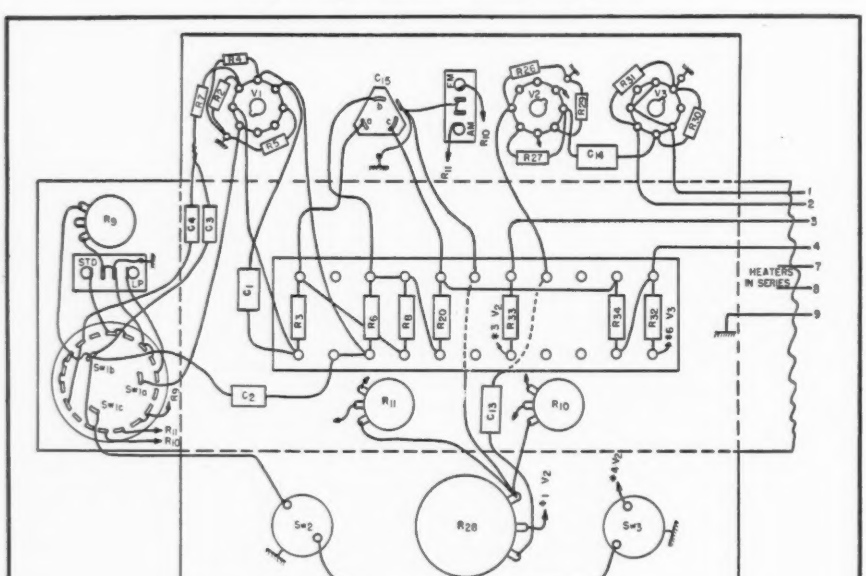
Fig. 4. Schematic of input section, with component values.

will be in the vicinity of .002 µf. *Figure 1* is a partial wiring diagram of parts employed in the power section, while *Fig. 2* is a photograph of the underside of the completed amplifier chassis.

The preamplifier section shown in the photograph of Part 1 is somewhat smaller than the average constructor may wish to employ, but it was built in this manner to keep the space required to a minimum. The chassis was fabricated from a 2 x 7 x 11 aluminum chassis base, cutting it apart at the center. The sides of the chassis are then cut 1¾ in. from the open end, and folded in to provide a ⅜-in. angle. The top is folded down, resulting in a chassis approximately 3½ x 7 x 2. The tube sockets and the electrolytic capacitor are mounted on this section, with the controls on the opposite side of the chassis. The input selector switch is mounted on one end, with a lever extending through the panel for its operation. If the chassis length were extended to eight or nine inches, the selector switch could then be mounted on the front in line with the other controls, and it is quite probable that it would be easier to install in a cabinet because of the difficulty in cutting a neat slot to pass the lever-type arm used to actuate the selector switch.

In the parts list several references were made to the text for further explanation. The volume control, R_{28}, is the loudness control described in the February issue,[1] and consists of a Centralab 1443 switch on which are mounted the resistors and capacitors necessary to obtain the desired compensation.

Fig. 5. Input section wiring diagram.



The high-frequency tone control switch, SW_2, needs a little further description. The series of switches selected for the tone controls consists of small units which are desirable in such a compact amplifier. However, the exact assembly of contacts is not obtainable, so the switch used was made by using the frame and mechanism from a 3115J switch with the contact plate from a 3126J switch. This gives a number of tie points for the resistors used for the cut-off circuit, as well as for the input connection. The capacitors in this circuit are all mounted directly on the switch, being soldered to the frame for ground connection. This is not usually considered the best practice, but it must be remembered that this section of the amplifier does not have any a-c circuits in it, and there is little chance of ground loops causing hum trouble. Suffice that the unit as constructed exhibits no troubles from this source. The exact arrangement of the tone controls is shown in *Fig. 3*. Similar methods were used for both, in that all parts are wired directly to the switch, but the low-frequency control is a standard 3115J switch without modification.

The overall schematic, *Fig. 4*, is essentially a repetition of the schematic published last month, with the removal of R_{12} and C_5 as previously discussed. The wiring diagram, *Fig. 5*, indicates the arrangement of parts and the wiring between them, while

[1] "Full Range Loudness Control," Winslow, AUDIO ENGINEERING, Feb. 1949.

the photograph of *Fig. 6* shows the method of mounting the resistor strip on the back of the volume control switch.

Unless the builder is reasonably well experienced in construction of small amplifier equipment, it might be desirable to increase the overall size of the input section. It is definitely possible to construct the unit in the size shown, but it must be admitted that it is extremely compact, and as the size is reduced the complexity of construction is increased. Of course, once the amplifier is completed there should be no need to get at it again, assuming that the constructor takes reasonable care in the selection of resistor and capacitor ratings to prevent the possibility of failure in use.

A more conventional construction of this entire amplifier would undoubtedly reduce the hum level still further, and if a larger power transformer were used—one which could supply the 0.9 amps of filament current required by three 6-volt tubes in the input section—the electrostatic field existing between heater and cathode of the present first 12SL7 would be reduced. If made in two units, however, it would be preferable to use a separate cable from the power section to the preamplifier to carry the heater current, although the signal from the preamplifier to the power section is of a relatively high level and it is possible that no trouble would be encountered from this source.

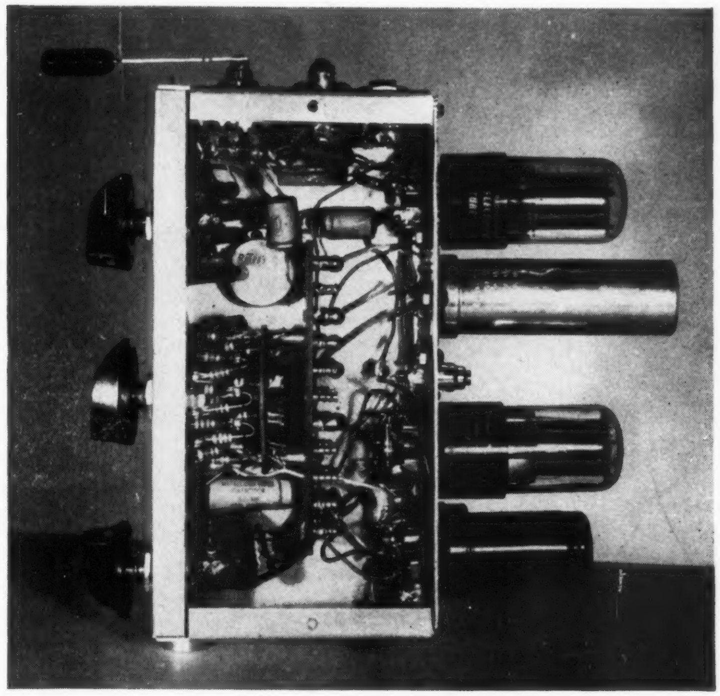


Fig. 6. Interior of input section, showing the mounting of resistor strip on rear of compensated volume control.

If additional gain is required, the cathode follower section of V_2 can be changed to a conventional amplifier, thus giving approximately 32 db more gain than with the present arrangement. For the uses for which this amplifier was designed, however, this should not be necessary. Another possibility is that a 6SN7 could be substituted for the 6SL7 used as V_2, (if a filament transformer were being used, together with 6-volt tubes) again using the amplifier connection rather than the cathode follower, and the increased gain would be of the order of 10 db. This suggestion would only apply if the filaments were arranged to be supplied from a transformer winding, since the 12SN7 will not operate in a series string with the 12SL7 and the 12SJ7, inasmuch as the 12SN7 draws a filament current of 0.3 amps. These are design modifications, and some ingenuity on the part of the constructor will be necessary to arrive at the exact desired result. It is felt, however, that the amplifier as described performs satisfactorily, and that no changes are necessary *for the purpose for which it was designed.* The amplifier was designed for one application—that of modernizing an existing installation, or for providing a control arrangement which could be adapted to cramped quarters with the greatest of ease. This requirement is fulfilled adequately by the arrangement shown.

Measuring Procedures for Magnetic Recording

ABOUT three years ago the Radio Manufacturers Association organized a subcommittee of the Committee on Phonograph Combinations and Home Recording to suggest standards for magnetic recording equipment. A series of sub-subcommittees were formed shortly thereafter, one of which was commissioned to work on "magnetic recording terminology" and to propose measuring procedures for magnetic recording. This sub-subcommittee has now released recommendations for measuring procedures for magnetic recording.

While it fully recognized that this material will be subject to modification with additional knowledge becoming available, it is felt that it presents a valuable basis for comparing the performance of various types of magnetic home recording equipment. To give this tentative proposal as wide a circulation as possible, the RMA Subcommittee on Magnetic Recording has approved its publication at this time.

Particular attention is called to the section on Noise which is subdivided into System Noise and Medium Noise. This subdivision has been suggested since in most cases the inherent noise generated by a completely neutralized medium is lower than the noise which might be expected in a typical magnetic recorder. It should be kept in mind, however, that noise measurements made, even on a properly neutralized medium, can produce indefinite results. To illustrate this point, a slightly magnetized reproducing head can greatly affect the noise generated by the magnetic recording medium and ambiguous results might be obtained.

For the proper evaluation of a medium, not only the ground noise of the medium as such is important but also the modulation noise (noise-behind-the-signal). No recommendations are presently available for an acceptable procedure for determining the noise-behind-the-signal. There can, however, be no question that there is a gap in the measuring procedures which eventually has to be bridged.

Another problem with which the investigator of magnetic recording equipment is confronted is the determination of amplitude variations which are experienced in reproduction when steady tone signals have previously been recorded. The performance of a magnetic recorder depends greatly upon the magnitude of such amplitude variations. Measuring procedures should eventually be provided to evaluate this deficiency of mediums and of complete magnetic recording systems.

Measuring Procedures for Magnetic Recording

RMA Sub-subcommittee on Nomenclature and Measurements
RMA Subcommittee on Magnetic Recorders R7.4

1. FREQUENCY RESPONSE OF UNEQUALIZED SYSTEM

The frequency response of a magnetic recording system depends upon the medi-

(Continued on page 41)

Lyle Van, WOR-Mutual's ace news commentator (right) hears principles of Altec miniature microphone explained by Paul S. Veneklasen (left), Altec Lansing physicist, who with W. J. Moreland worked on 3-year development project of the tiny microphone. Lyle Van's six-o'clock news broadcast Tuesday, March 8, was first time the new microphone was used on the air.

JOHN K. HILLIARD*

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An Omnidirectional Microphone

The entire communication field advances to higher standards when previously unconquerable obstacles are surmounted. An indication that such a turn of events has occurred in the audio field comes with the introduction of a miniature condenser microphone, described in this article, with a diaphragm having the area of a human ear drum.

A large number of microphones have been developed since the original work of Alexander Graham Bell. In the broadcasting, recording, and sound reinforcement fields, the double-button carbon microphone was the earliest to come into widespread use. This was later replaced by the Wente type condenser microphone in 1927. A few years later, dynamic and ribbon microphones replaced the large diameter condenser type so as to obtain more uniform response and simplified equipment. Later, directional microphones incorporating features of both dynamic and ribbon types became available and were designed to provide various response patterns such as a figure eight (bi-directional), cardioid and modified cardioid patterns. Theoretical patterns indicate that ideal conditions can be achieved only under "dead room" conditions. These patterns are then considerably modified when used indoors, due to reflection from the wall surfaces. In the extreme case of very reverberant rooms, little directional gain can be obtained over that of non-directional microphones. Also, directional microphones are larger than non-directional units and the sound field is accordingly distorted by their obstacle effect. This results in a directional pattern that varies over wide limits in frequency. The response angle becomes smaller at the higher frequencies.

Obstacle interference is a direct function of size and frequency. This varies the directivity and absolute output of a microphone. The solution to the problem lies in making the size extremely small.

A high degree of cancellation in cardioid microphones is usually obtained only at the middle frequencies. Compromises must be taken in phase shift by the use of mechanical and electrical networks to obtain approximately 180-degree cancellation. Earlier experiments and data on phase shift indicated that the human ear has difficulty in distinguishing between a system with considerable phase distortion and one without. However, later experiments indicate that a minimum phase shift is highly desirable in high-quality systems. In a comparison of high-quality systems having small and large amounts of phase distortion, the difference can be detected by a critical listening test.

All the microphones mentioned which have been available for high quality work have many undesirable characteristics, the most important of which are listed below:

1. Their large bulk causes shadows in Motion Picture and Television work requiring that they be moved farther away from cast than otherwise desirable. In direct broadcasting or stage presentations they hide the face of the performer, since usual technique requires that the artist work very close to them.
2. Bass tones are unduly emphasized as the artists work closer to the microphones. This results in boomy reproduction with poor intelligibility.
3. Microphones having freely suspended ribbons are susceptible to puffs of air caused by breathing of the artist. This causes an objectionable rumble and in the extreme can produce damage. They are not shock-proof—a loud hand clap, or gun shot can render them inoperative. It is difficult to use them for exterior work in the presence of wind.
4. Because of their directional characteristics at close range, many microphones are used to give a required musical balance in orchestra or band pickup.
5. They have powerful magnets which attract iron filings and often block the microphone during operation.
6. The directional microphones limit the reproduction of the natural room tones which have been carefully designed by the architect for pleasing listening.

In order to overcome many of these undesirable features and limitations, a miniature condenser type microphone has been designed. Its overall dimension is 6/10" in diameter and 4/10" thick (approximately the size of a stack of six dimes). It weighs less the ¼ ounce (6 grams). A small circumferential sound entrance channel

20 mils thick provides protection for the diaphragm and aids in obtaining an omnidirectional pickup characteristic. The diaphragm is one centimeter in diameter and is of a special laminated construction. The resonance of the diaphragm is at a high frequency and its peak is controlled by the damping so that it is negligible, varying ±1 db from its low-frequency sensitivity.

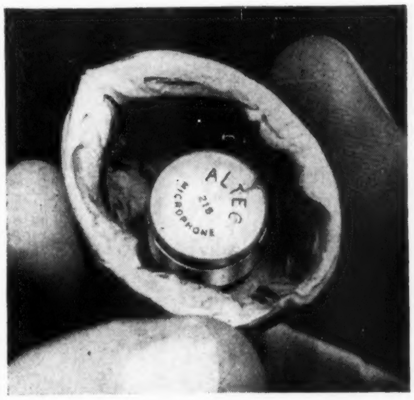
The microphone is mounted on a base which has a maximum diameter of one inch. This base contains a 6AU6 miniature vacuum tube which is easily replaced and the bottom of the base contains a Cannon 6-pin plug.

The stand mount and output cable contains a Cannon receptacle and the necessary fitting with a ⅝" -27 thread for stand mounts. This cable is in effect a single-conductor shielded cable designed to perform the necessary functions of the impedance-transforming tube by dividing between the central conductor group and an outer group which also functions to shield the inner signal carrying group. This outer shield provides the leads for heater current, high voltage supply and ground from the power supply. This cable may be as long as 400 feet and is attached by a 6-pin Cannon plug to the power supply and matching transformer unit.

The output level of the microphone system is —50 dbm in a sound field of 10 dynes/cm.[2] The matching transformer provides output impedances to work into equipment having assigned input impedances of 30-50, 150-250, 500-600 ohms. The output noise level is less than 30 phons.

The cable which supplies the impedance-transforming tube is in effect a single conductor cable in which the central signal-carrying conductors are surrounded and shielded by those which supply the tube operating power.

The blast-proof characteristics of the microphone make it unnecessary to protect it from extreme sound pres-



The miniature condenser type microphone, photographed inside a walnut shell, is shown here actual size.

sure levels and shocks which might cause distortion or damage to other type microphones. Tests made on this condenser microphone by firing a 22-calibre cartridge at a distance of eight inches indicate that its calibration is unchanged by such a blast. Based upon electrostatic rather than electromagnetic principles, it is not susceptible to iron filing damage or magnetic induction. The size of the housing and the diaphragm construction allow the microphone to be used under wind conditions in external pickup work that otherwise would be impractical. In public address and sound reinforcement applications, it is possible with this microphone to achieve at least 4 db higher amplification before feedback or howl is encountered than can be obtained with other non-directional microphones equal to the best directional microphones. Flat response (lack of peaks) explains why this is possible—plus the fact that the sound is practically all reflected energy when more than 20 feet away from the source (in rooms).

The microphone base is separated from the microphone by an extension which serves to reduce the obstacle interference in the immediate field of the microphone and also permits the microphone to be used directly in front of the artist with little shadow.

The weight of the combined microphone, base and extension is approximately four ounces. It is anticipated that because of its extremely small size, weight and shadow as well as other outstanding characteristics mentioned above, this microphone will facilitate heretofore unachieved pick-up in the radio, television, recording and sound reinforcement fields.

The microphone system is the first of a series of apparatus resulting from a new basic investigation of sound pick-up limitations. Major changes in technique, the development of new uses, and correlative advances in other fields are expected to result from this development.

So many Altec engineers have contributed to the development of this new microphone that no one person can be singled out for special mention. This project was initiated and supervised by G. L. Carrington. P. S. Veneklasen and W. J. Moreland were responsible for important electronic and mechanical developments. Capacitance of the microphone unit is less than 20 micromicrofarads.

Mass Production Tape Recording

The final technical obstacle in the way of mass production of recorded music on tape now has been overcome. A machine has been perfected by Minnesota Mining and Manufacturing Company, St. Paul, that can simultaneously reproduce 48, hour-long tape recordings indistinguishable from the master transcription in one hour, according to an announcement by W. L. McKnight, president of the firm.

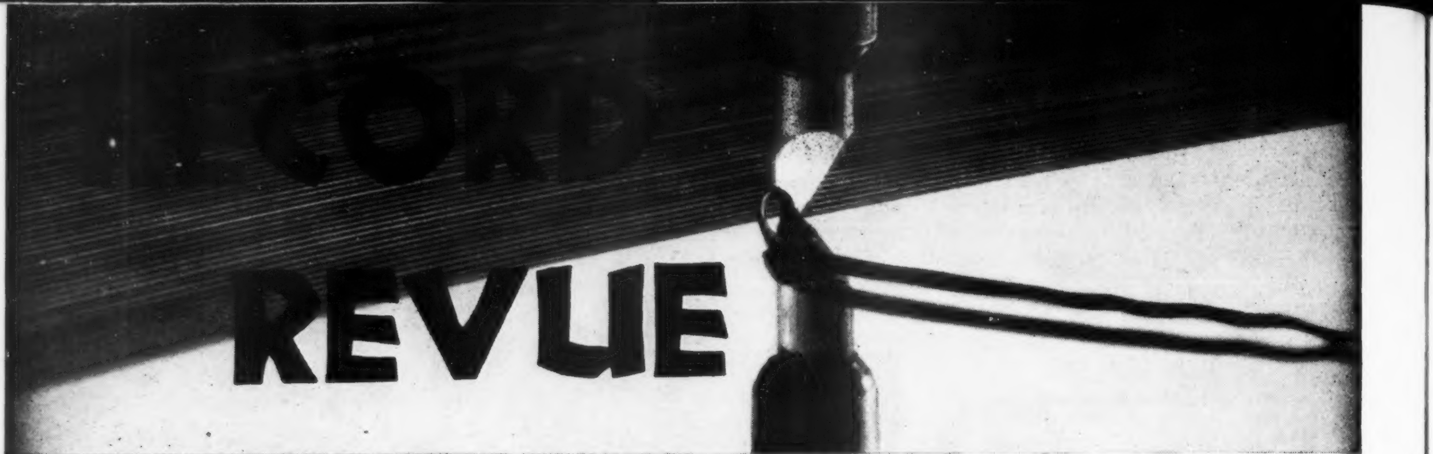
These pre-recorded reels of tape will be designed to compete with disc records for use in the home, in broadcasting, in schools and theatres. Since many sound engineers contend that magnetic sound tape has better fidelity than any other known sound recording medium, it is to be expected that recorded music on tape may enjoy a competitive advantage over disc recordings, whether of the 78, 45, or 33-1/3 rpm variety. Tape recording machines for home use have been available for some time.

Minnesota Mining and Manufacturing Company describe the new machine as "a high fidelity multiple recorder capable of making tape recordings which are indistinguishable from the master transcription." Reproduction is accomplished by an electrical duplicating process in which the signals from a master copy are picked up by a playback head, amplified and fed electrically into a number of re-recording heads. The master tape and the tapes to be copied are all run side by side on a common capstan. Thus exact speed relation is maintained at all times.

Recording tape is currently available only in the unrecorded form, which is used by the major broadcasting networks as well as in homes, offices and schools for making private transcriptions. Announcement of the multiple recorder, however, opens the way for mass production of pre-recorded music and promises to affect the whole field of sound recording. Perfection of the multiple tape recorder follows hard upon the introduction of 45 and 33-1/3 r.p.m. disc records and machines by Victor and Columbia respectively.

The single mass production recorder built to date has been offered for lease by Minnesota Mining and Manufacturing Company, its producer.

[Continued on page 47]



EDWARD TATNALL CANBY*

OUR EDITOR has brought up a point concerning electronic musical instruments where I was technically in error, though the point I had to make was unaffected. In the February article I spoke of "pure" tones to be heard on an electric organ; it seems that according to engineering standards of measurement the fundamental tones produced by such devices are not pure, that indeed a pure tone is very difficult to achieve. What a pure tone is, then, is a matter of engineering vs. musical terminology. From a musical viewpoint, any tone that comes even fairly close to a continuous sine wave form is for all intents and purposes "pure"—by which we musicians mean, of course, something highly uncomplimentary. A continuous "pure" tone in music is a dead duck.[1] It's lifeless, featureless, without character, unfit for musical purposes altogether—just a noise. I'd say the principal musical difficulty with the electronic musical instrument is still that of oversimplification of the wave forms. You will find that the continuing distrust of such instruments (often a highly unreasonable blanket distrust that includes various instruments that work on utterly different principles) is, nevertheless, basically a feeling that the gadgets just aren't musically "natural," and this feeling must be reckoned with.

And by "natural," I hastily add, we do not mean that your electronic sound must be a good imitation of some specific natural sound. The advantage of the electronic sound is that it can introduce all sorts of new qualities. Few musicians would quibble about this—unless provoked by manufacturers who insist upon using such terms as "electronic clarinet." We are entirely ready to recognize the musical value of new tone color combinations whatever names may be assigned to them; silly, fanciful, inspirational, or what.

But whether your newly created electronic sound comes forth like a cello or an oboe, a new-born babe with asthma or a honking goose (a goose is straight-bore, producing only the odd overtones) the musician will give ear and instantly make up his mind on one vital point—whether the tone is *alive* or *dead*; musically natural or unnatural. Being no engineer, your musician will seldom be able to explain what he means. Being only human, he will give you lots of reasons for his energetic (and probably correct) opinions that will make no engineering sense, for the most part. (Note the singing teacher who insists that her pupils generate their tone somewhere in space a few feet in front of the forehead! Cockeyed, but pragmatic; it works, because with such conceptions singers actually do learn to produce good sounds.) Explanations in physical terms of these things are up to the engineers to make. And in every case the musician, however false his physics may be, has a basically sound idea, needing only the delicate physical interpretation that will reproduce it in electrical terms. What an astonishingly difficult and challenging area this fringe-area is, between the arts and the sciences, between objective scientific analysis and the far quicker, more penetrating human intuition! In the end, as we all have to admit, it's usually "hunch," that solves the most baffling problems, when the "hunch" is well backed by competent engineering training.

What t h e n, is an "alive" musical tone? How can it be synthesized? This column has touched on the matter before, but look at it this way. To be musical a tone must be complex, irregular, and that in many different ways. It must avoid all sorts of too-regular mathematical patterns that will kill its musical effect. A paradox, since most of us have supposed that pitch and rhythm in music were at least reasonably fixed quanitities, not to mention tone color. But no. Life, in tone, is highly delicate; it succumbs instantly to an overdose of regularity. Yes, there must be a sense of pitch, a fundamental tone that is interpretable to the ear in a very exact sense. But even this is seldom mathematically exact. A tone that is interpretable to the ear as an excellent musical "A" may be far removed from a pure 440 tone. Almost all musical tone, for example, includes the baffling feature of the vibrato. It is frequency modulated. Take a lady opera singer producing a walloping A and play her 78 rpm record at 33. You will be astonished at what the ear has managed in its own strange way to interpret as a pitch of 440! The vibrato may waver up and down so enormously that, at 33, the ear recognizes no fixed pitch whatever but only a sort of rapid-action fire siren. A graph of the sound would be even more baffling. Yet this isn't all—for even the vibrato's own frequency must not be fixed, if the tone is to be musical—as electric organ builders know to their own distress. The vibrato is *almost* regular, but never exactly so—make it mathematically regular, and instantly the tone is dead.

Moreover, the amplitude of your vibrato must not be exactly regular. Amplitude too must vary minutely from instant to instant as the musical tone continues. Fix the amplitude mathematically and your tone is dead. Listen again to your lady singer and note how much the varying amplitude has to do with the musical sense of what she is singing—some passages being almost "flat," others being highly modulated with vibrato. So too with an oboe, a violin, even a trumpet. Only a few instruments dispense with the vibrato: the piano of course and its relatives; to a considerable extent, the clarinet.

But vibrato is merely one factor that must avoid the mathematically regular. Even more vital is the tone-color problem. Engineers tend to take an *average* tone color for, say, a clarinet, and assume that this is that instrument's official tone color. There is scarcely any instrument which produces a constant overtone pattern for more than a fraction of a second at a time; one of the main sources of musical "life" is the dynamically changing tone color of nearly all instruments as they (*a*) change pitch and (*b*) change intensity and (*c*) change methods of tone production. Perhaps the natural pipe-organ is the only instrument with a fixed overtone pattern that actually persists measurably.

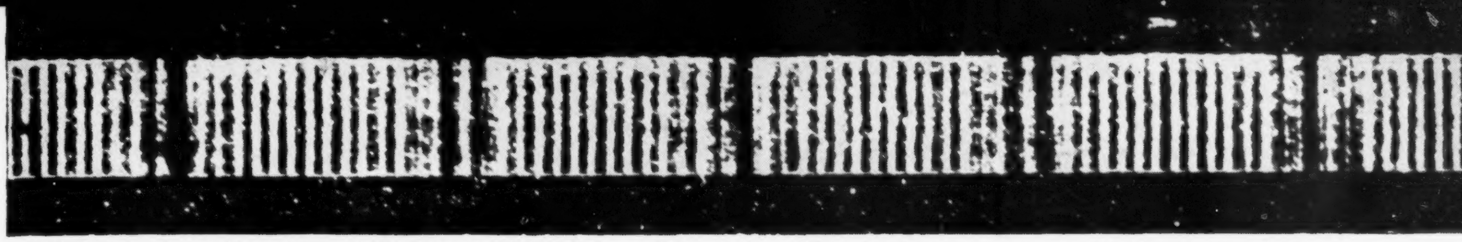
All of which no doubt leaves most engineers exasperated—if music is as touchy as all this, what in Heaven's name is the use of trying to please a musician! And yet this *is* music and always will be. Exasperated or no, these are the actual problems that a sound engineer faces in any electronic musical instrument design and they cannot be dodged, only compromised with. When you come down to it (and this a fine reason for this column's discussion of the problem) the same sort of thing is true for the recorded art. If you are going to reproduce *musical* sound in a natural-sounding way you must meet the challenge of the musical ear. Your success as a sound man, given sound training, will depend on it, too.

Any electric organ designer knows that problems of naturalness go far beyond what I have suggested here. One must get the tone under way with the proper "ping" (the musical ear will take nothing else) and one must get it properly stopped, too. Can't just turn it on and off. One must see that the tone "speaks" a bit late in the keyboard-type instrument; the musician won't like an instantaneous action. Unnatural. Theoretically, as volume increases tone color should become more brilliant. Natural music works that way. And so on. These problems bedevil sound engineers and in practice they are bound to involve very great compromises all along the line, depending, of course, on the cash available to put into refinements. Theoretically anything is possible. Practically, the most ingenious compromise wins out musically. Far too often in today's electronic musical instrument some regularity gets in that "kills" the tone as far as

[*Continued on page 33*]

**279 W. 4th St., New York 14, N. Y.*

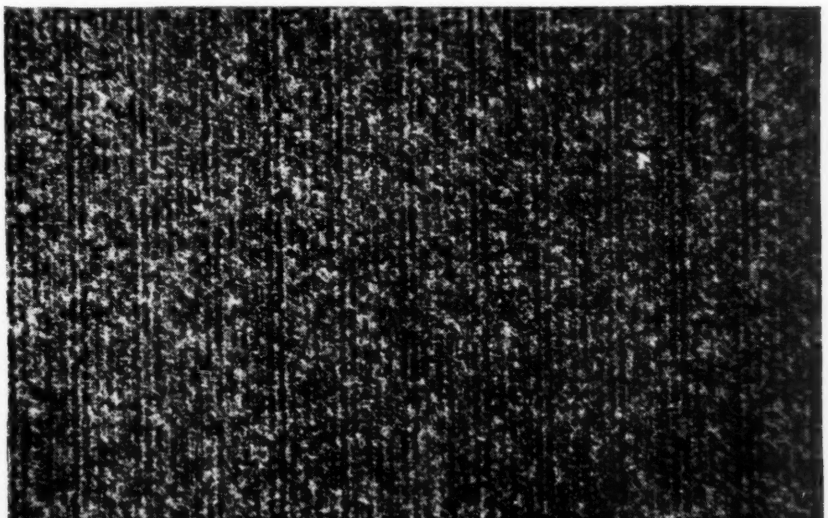
1 The flute, as engineers know, can give an almost pure sine wave tone. But note well that such a tone is hardly what one hears in flute music. A flute player making actual music produces the same multiple varieties of irregularity that are described later in this article. There must be the ictus of the initial accent, a brilliance in loud passages (harmonic tone color) contrasting with less brilliance in soft passages, a vibrato that is among the most noticeable (and most irregular) in any musical instrument, a breath-sound that is normal part of the music (and is not a sine wave)—all this and more, in addition to rhythm, phrasing, etc. A mechanically played flute, actuated by a constant stream of compressed air, may produce a sine-wave—but the sound will be noise, not music. No electronic sine wave "beep" has yet been compared to flute music, that I know of!



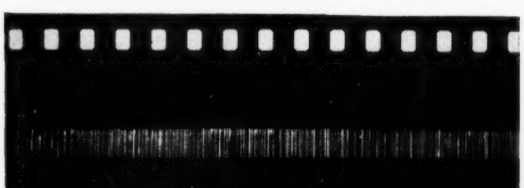
The beat note resulting from a high level recording of the two frequencies, 200 and 230 cps, is seen in this five times enlargement.

Making Magnetic Recordings Visible

The technique used in making visible the sound tracks shown on this page was described in an article entitled, "Alignment of Magnetic Recording Heads" by B. F. Murphey and H. K. Smith in the January 1949 issue of AUDIO ENGINEERING. For some purposes, where several inches or feet of tape are to be visibly examined (as for editing), Mr. Robert Herr of Minnesota Mining & Mfg. Co., who has supplied these pictures, reports a more convenient and less messy method. The carbonyl iron is suspended by shaking in a volatile liquid, such as heptane (which will not dissolve the tape) and the tape is dipped in this suspension for a few seconds. Upon removal, the liquid will dry quickly and the track becomes visible. The carbonyl iron may be removed by wiping it off. This method allows some flocculation of the particles and does not yield quite so good resolution as the suspension in a more viscous medium, but it is simpler and adequate for examination by the naked eye.



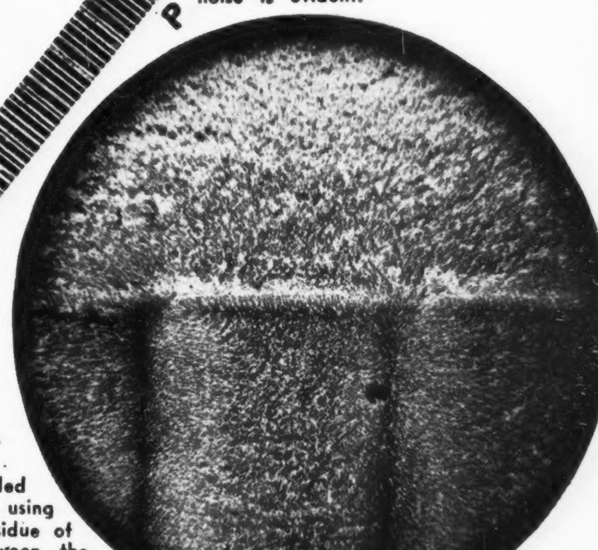
Above: A constant tone modulated by a vibrating head is shown here. Magnification X 60.



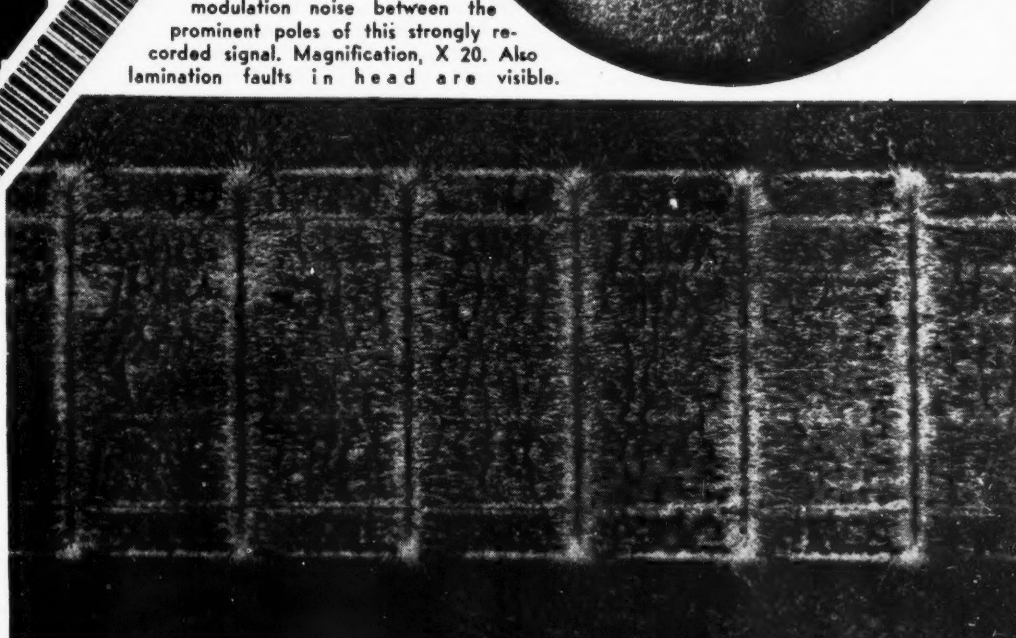
Above: Music recorded on oxide-coated 35 mm film is illustrated by this photo. Ready means for editing and track location is provided by making the track visible. No enlargement.

Below: The word "tape" was recorded with a full width 1/4" track, using an Ampex machine at 30 inches per second. Enlargement, X 1 1/4.

Below: 26-times enlargement of a 0.1-inch wavelength signal recorded on black oxide tape illustrates the fringing effect. In contrast to the other photo of a 100-mil track, no modulation noise is evident.



Below: A 0.1-inch wave-length recorded on a 0.1 inch track using noisy tape shows residue of modulation noise between the prominent poles of this strongly recorded signal. Magnification, X 20. Also lamination faults in head are visible.



Experimental Ultrasonics

S. YOUNG WHITE*

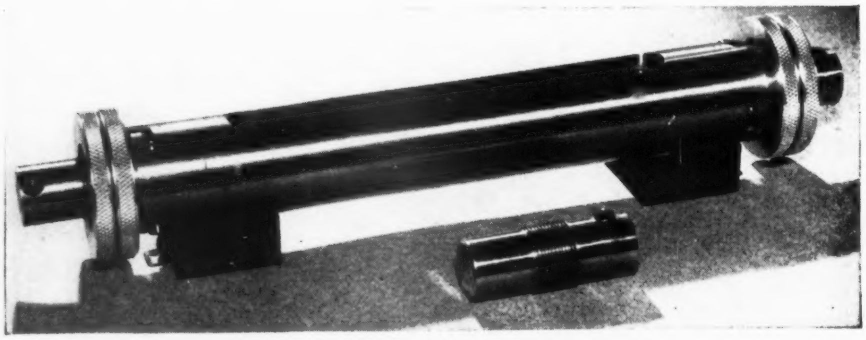
Part II—A description of some of the problems of instrumentation for experimental work in the realm of ultrasonic frequencies.

ABOUT THE MINIMUM AMOUNT of equipment required in the investigation of ultrasonics is a source of ultrasonic power, such as the Hartmann generator, a microphone or pick-up, and an indicating device such as a receiver or oscilloscope. Of course, other indicators can be used, such as pigeons or insects, which have positive reactions of various kinds to certain frequencies and amplitudes, but the lack of some measurement devices is rather a handicap.

One great difficulty in attempting to turn out a standard line of ultrasonic apparatus is the very complexity of the requirements. A biologist might wish to investigate the effect of ultrasonic stimulation on cancer, and a cathode-ray tube production man may wish to use the well-known dust precipitation effect to more rapidly coat his screen with phosphorus. In these cases both the generators and observation instrumentation would be quite different. Also, the biologist would be rather unskilled in using apparatus of this nature.

In this article we shall discuss some design features of rather small pick-up devices, such as would be used in a probe. It is almost universally true that all these receiving probes can just as well be used as low-power transmitters, so it would be a great advantage always to design them for both functions. It is also desirable to make them immersion-proof, if possible, so they can be used in liquids as well as air or other gases. Most of them have a natural temperature limitation of some kind, also, and we wish to extend this to its highest limit possible.

Another requirement is small size. These are to be used in sonic fields of small wave-length, and in general we wish them to disturb the field as little as possible. Of course, if we work up around the megacycle region, the physical size must be many wavelengths in dimension for practical apparatus for general use, although special microscopic devices



Test chamber used with two transducers for experiments with transmission through various media. One of the transducers of Fig. 1 is shown separately.

can be made if the requirement justifies the expenditure.

Shielding against powerful electromagnetic or electrostatic fields is often necessary. As a rule, 60-cycle hum pickup is small, but can be troublesome if not kept in mind. A particularly difficult case is when two magnetostrictive probes are used close together as transmitter and receiver. Very often the flux leakage from the transmitter will directly excite the receiving unit and the supersonic coupling through the medium will be masked.

It is a great advantage to design a series of units interconnected with modern, 52-ohm concentric cable. It has a nice line of fittings available for all purposes, provides good shielding, is low-loss at all frequencies we may wish to use, and is very neat and durable. Unfortunately, about the only type unit that can readily be made to match this low impedance is a magnetostrictive one. All the piezoelectrics suffer severely from the high capacitance, and sometimes it is necessary to go to troublesome or expensive steps to overcome this.

A real hardship in designing a line of units is the wide frequency range we generally wish to cover in general experimentation. A final commercial installation can often be designed to operate at one frequency only. Before we can determine this optimum frequency it may be necessary to investigate an enormous range of many octaves. In working with small particles in water it is usually interesting to investigate the entire spectrum from 10 kc to nearly 10 mc. This is a real test of the designer.

It is the purpose of this table to give a rather rough idea of the frequency limits of various transducers we may wish to use. The range designated as "natural" shows the frequency limits within which no unusual difficulty is had in normal design. The "extreme" range shows maximum performance ever observed by several experimenters. It may be only a single sharp peak shown in some unusual mode of oscillation.

Significance of the Resonant Peak

In general terms, we can operate our devices at resonance or off resonance by any desired amount. Since

TABLE 1

Device	Natural Range	Unloaded Q	Extreme Range	Max. Temp. (F)
Rochelle Bimorph	0 to 30 kc		0-2 mc	120
Rochelle single	0 to 300 kc	20,000	0-3 mc	120
PN	0- 200 (piston)		0-3 mc	250
Magnetostriction	10- 100 kc	30,000	200 kc	250
Rubber electrostatic	10- 500 kc		3 mc	150
Quartz	50- 7,000 kc	10,000	Very high	1,000

———

*% *Radio Magazines, Inc., 342 Madison Avenue, New York 17, N. Y.*

many show unloaded Q values that are very high, we can mount them so that high Q is maintained, or load them down to almost any extent.

The chief argument for high Q is the great output obtained at that resonant peak, a very strong argument indeed. The price one pays for it is high, however.

If the units are operated in pairs, one being the loudspeaker and the other the microphone, this resonance rise is squared, and the output is very favorable. The frequency stability on the transmitting signal source must be very great indeed, and in general is hardly practical. If we use a precision frequency standard for the signal to drive the transmitting transducer, it will not follow the transducer peak in many cases, as the transducer is out on the firing line, subjected to variations in temperature, vibration, pressure changes and the like while the standard crystal is locked up in a temperature oven. Also, in the field standing waves with ratios of ten to a hundred are rather common, and these change due to local conditions.

This severely limits the use of the high-Q transducer. So in practice a load sufficient to bring the Q down to as low as ten or thereabouts is often employed. Even at this low Q standing waves are severe, and frequency modulation must be employed to break them up.

This loading does not seriously affect the off-frequency operation of the devices, but of course operation off frequency severely cuts down the signal. A typical example is a pair of one-quarter inch cubes of PN crystal suitable for stable operation in a water supply which might have particles in it. At 200 kc operation, resonance for both units, an input of 200 volts of signal on one will give perhaps 2 volts on the receiving crystal, while off resonance at say 150 kc the combination might give an output of 5 millivolts or so.

Diaphragm Design

The only transducer diaphragm easy to design is the magnetostriction type. It is inherently rugged and needs little protection from gas or liquids. A simple metal disk silver-soldered on the end of the nickel rod, gives almost any desired acoustic impedance match.

PN and Rochelle crystals are very fragile and must be protected from shock and damage if used as contact microphones. PN instantly melts if one drop of water comes in contact with it, but Rochelle salt often can be given a protective coating to prevent this. Quartz has high resistance and moisture must be kept out to preserve the insulation of the assembly. A 7-mc quartz crystal, of the X-cut type is about 20 mils thick, and is quite fragile.

So we have our choice of rubber (plastic) diaphragms, or solid metal. The rubber must be quite thin and offers little protection from sharp points. However, diaphragms made from thin rubber, nylon, or Teflon of a thickness of only two mils or so are very easy to handle, as below 200 kc or so they have little effect on the transmission of ultrasonics through them. To provide clean-cut effects, they must be placed under tension in some way, and the mounting made waterproof by some means.

Standardized Shell For All Frequencies

Figure 1 and the photograph show an earnest attempt to design a universal shell of wide application in ultrasonic commercial and experimental work. In most cases it has a metal diaphragm. This was designed to operate through the spectrum from 10 kc to 7 mc, using practically any type of transducer from magnetostriction to X-cut quartz.

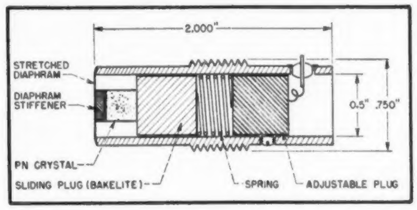


Fig. 1. Universal shell design for transducer suitable for wide range of ultrasonic frequencies.

An accurate half-inch bore dimension was used with a thick wall so that set-screws and glass-bead lead-throughs could be used. A three-quarter inch thread was formed on the center section and this made the tube outside diameter 0.690 inch, as this cleared the root diameter of the thread. This thread was made rather fine, 24 threads/inch, as very often we want fine adjustment of the position of the diaphragm in our tests. The shell was usually made of monel, because it is an attractive material unaffected by many liquids. Sometimes it was hard chrome-plated, or gold plating was called for. A length of 2 inches worked out well.

The series of diaphragms gave us the most trouble to design. If we use quartz or PN crystals, they are mechanically dead flat, and the diaphragm is often used under at least city water pressure, which might reach 60 pounds/square inch at times.

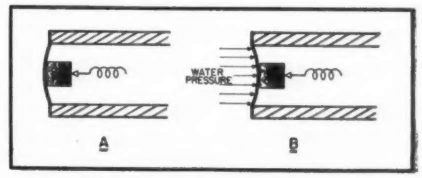


Fig. 2.(B) When used under pressure, diaphragm is pushed inward, providing contact on only a portion of the crystal surface. (A)Pressure behind the crystal bulges diaphragm outward, provides contact only on corners.

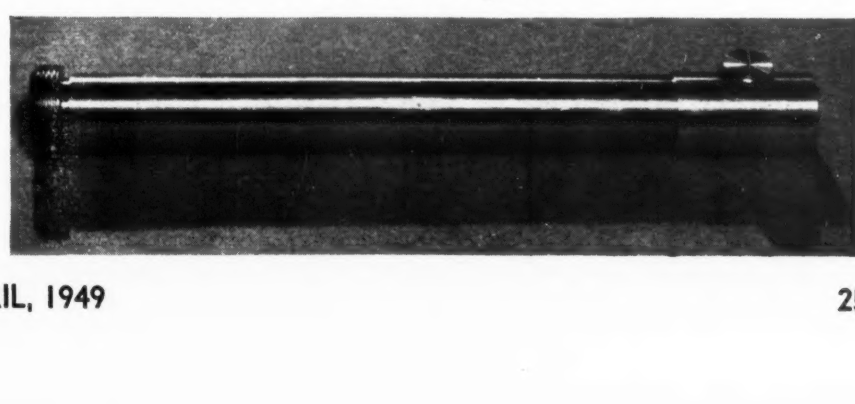
This bulges the diaphragm inwards, as shown in *Fig. 2B,* and the crystal would only touch in the center of its working face. In general, we need some pressure driving the crystal against the diaphragm, to obtain intimate contact, and with no pressure on the outside of the diaphragm a reverse bulge takes place, as in *Fig. 2A*, and the crystal touches on the corners only.

By soldering a one-quarter inch button to the center of the diaphragm, we accomplish two objectives: At some high frequency any diaphragm breaks up into complicated modes of oscillation, which produce lobes in the radiated or received pattern. A piston action is much easier to analyze and use. This local thickening in the center turns the quarter-inch center into a piston very well, as experience has shown, and gives a pattern very easy to use.

In practice, some care must be used in making and mounting this center button. Since it will often be used with a precision finished transducer element, it must be dead flat and squarely mounted. If slightly cocked, the pattern put out is hard to analyze, and results are undisciplined.

[*Continued on page 38*]

Transducer probe for use in air. Details of the interior of the crystal mounting are shown in Fig. 3.



The Cutting Sylus Problem In Microgroove Recording

"Stylus"

A discussion of the effect of burnishing-facet dimensions on frequency response.

THE CURRENT INTEREST in new methods of sound recording has made many an engineer re-examine more critically the faults of older systems. Modern disc recordists have become increasingly conscious of diameter effects, and particularly of loss of high-frequency response at smaller diameters. This loss occurs in two parts: First, a loss in *recording* due to the cutting stylus shape; and secondly, a loss in *reproduction* due to the reproducing stylus failing to follow the finer groove convolutions faithfully (tracing loss). Reproducing loss can be reduced by using a smaller stylus tip, within reason. In the case of lacquer, it can be reduced by using a harder lacquer, again within reason.

Only recently has there been much interest in minimizing recording loss. It is not generally realized that com-

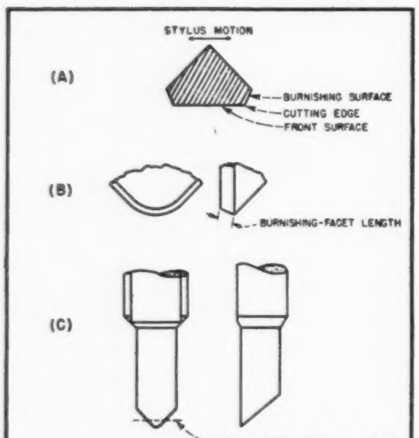


Fig. 1. (A) General view of lacquer cutting stylus. (B) Magnified view of cutting stylus tip. (C) Magnified cross section of tip.

monly used lacquer-cutting styli polish the groove walls by a burnishing action after cutting. This effect is comparable to the use of a dulled cutting tool in making a polished cut in brass.

As the polishing action is made more effective by increased burnishing-surface length, the high frequency response deteriorates. A glance at

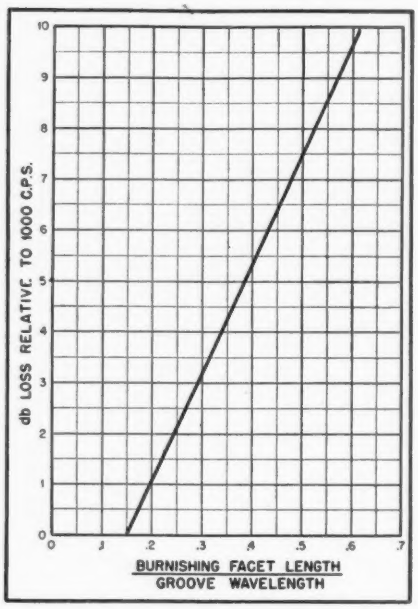


Fig. 2. Generalized effect of burnishing facet length.

Fig. 1 will illustrate why. Effectively, there are flats on the sides of the stylus, which impede lateral motion.

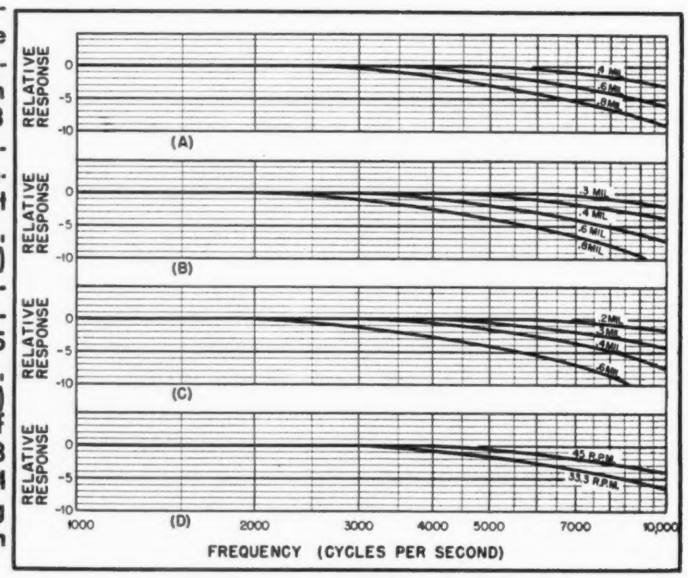
The shape of the stylus and the nature of recording lacquer are such that it is not feasible to derive a solution by theoretical means, and so laboratory methods must be used. The only quantitative study of this phenomenon, which has been published, is that of LeBel[1] in 1942. While the original study was made with a standard-size groove, we believe that its conclusions may be applied to the microgroove without too much error. Every cutting stylus has a certain amount of personality of its own anyhow, and the variability so produced is superimposed on the general trend.

We have used the generalized solution given by the above reference—a graph showing the relation between the fraction: burnish length/wavelength, and the attenuation in recording. We have taken the liberty of continuing the main trend for a db further than the reference has, for reasons that will be evident on inspection of his data. This slightly modified graph appears as *Fig.* 2.

The tests on which this curve is based were run at a stylus velocity of 1½ inches, and checked at 2⅝ inches per second. Visual inspection of *Fig.* 1(A) will suggest that the loss might increase at very high peak velocities, but the velocities indicated include the range of velocities normally expected.

The LeBel study was made in terms of *tip* burnish length, which is the easiest to measure accurately. The

Fig. 3. (A) Frequency response of recording stylus at 8 inch diameter, 33.3 rpm. (B) Frequency response of recording stylus at 7 inch diameter, 33.3 rpm. (C) Frequency response of recording stylus at 5 inch diameter, 33.3 rpm. (D) Comparison of response at 33.3 and 45 rpm, .4 mil burnishing facet, 5 inch diameter.



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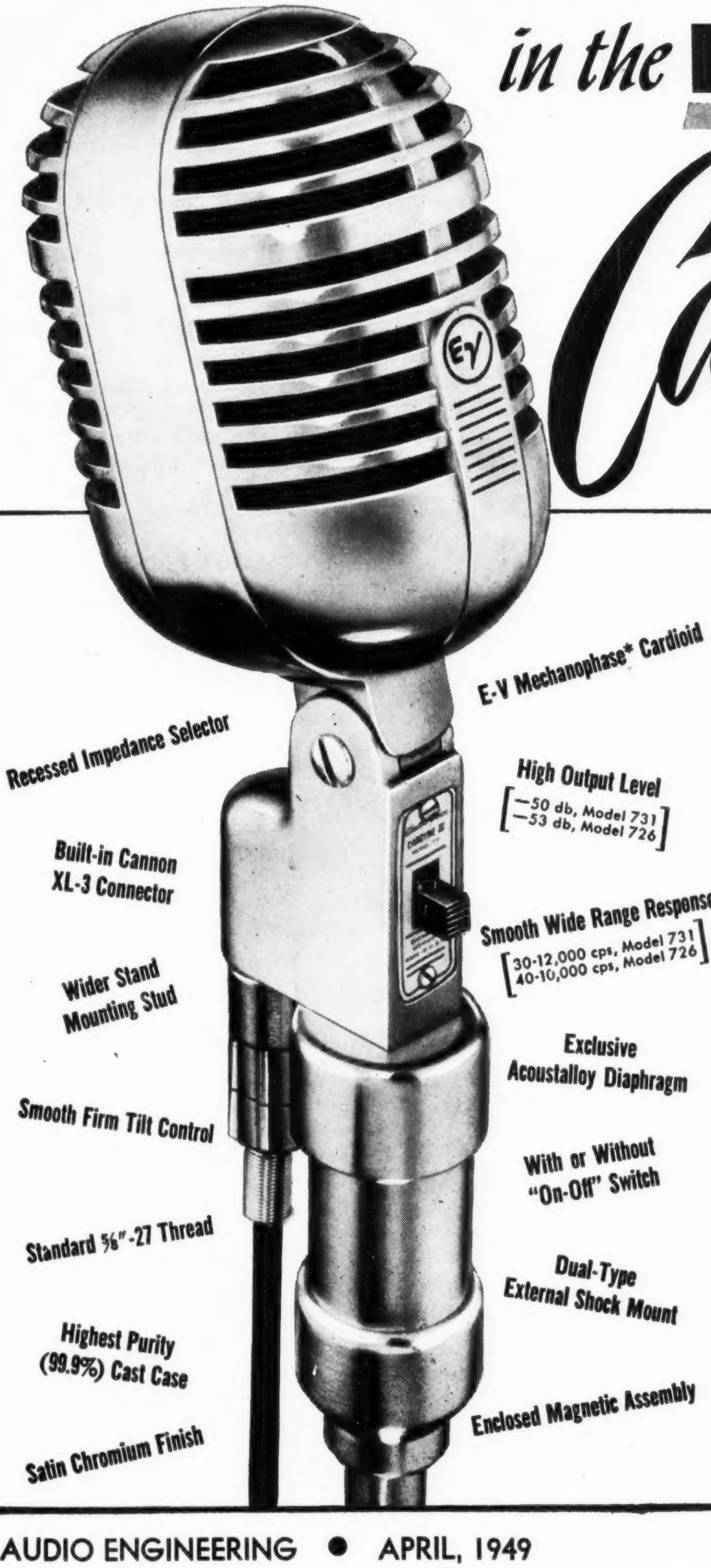

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length at the sides may be slightly less, with the decrease somewhat under the control of the lapidary.

The following formula is too obvious to require derivation:

$$\frac{\text{Burnish length}}{\text{Groove wavelength}} = \frac{f\,l}{\pi\,Dn}$$

where f = frequency in cps
l = burnish length, inches
D = recording diameter, inches
n = speed in rps

A glance will show that the fraction increases with frequency and burnish length, and decreases with an increase of speed or diameter. From this formula the graphs of *Fig.* 3 were computed.

A condition typical of standard high-quality 16-inch transcription recording is shown in *Fig.* 3(A). By the use of 136 or 144 pitch, the inner diameter is limited to 8 inches.

At (B) we have a condition occurring in many broadcast stations which have continued to record transcriptions with the pitch of ten years ago. This results in an inner diameter of 7 inches. The injurious effect of a given burnish is measurably greater, as can be seen by looking at the 10 kc response. Incidentally, this is fairly close to the outer recording diameter of a 7-inch microgroove disc.

Conclusions

Finally, we go in to the innermost diameter of a microgroove record—about 5 inches—at (C). A very small burnish is necessary, else the high frequency loss rises rapidly.

Just for comparison, at (D) we have plotted the response of a .4 mil burnish at 5 inches diameter at 33.3 and 45 rpm. The higher speed eases response problems, by 3 db at 10 kc in this case. This saving taken alone would not be significant, but similar savings accrue at other stages in the recording process, making a significant total.

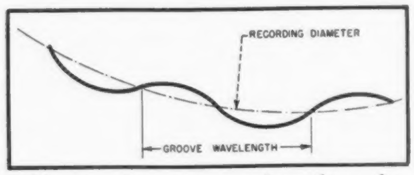


Fig. 4. Groove wavelength and recording diameter.

With current interest in higher fidelity, it is evident that burnish length will have to be limited. The high quality transcription recordist and the microgroove specialist will both have to use care.

Transcription work will have to use a burnish of not over a half mil, approximately. It is not difficult to make such a stylus satisfactorily quiet. A great many 10- and 12-inch microgroove discs are recorded in to only a 7-inch diameter, and a similar stylus will often suffice. On the other hand, the recordist working with 7-inch microgroove discs will be faced with a real problem. Compelled to use a burnish of about a quarter mil, he may have difficulty in getting a quiet groove unless he uses a higher cost stylus. It is a good deal harder to get a quiet groove when using a small burnish, and the lapidary may have to be much more careful than he is with a half-mil burnish.

In spite of all these precautions, not every stylus will have equally good frequency response, and it may be desirable for the recordist working at less than 6-inch diameter to check each stylus individually, using the poorer points for less critical work.

It should also be recognized that small-burnish styli of the best quality are not all equally quiet. Some are satisfactory all the way in to 5 inches, while others start to become noisy at the 6-inch diameter. It is highly desirable to segregate the latter for use at the larger diameters only. Again we may say that each stylus has a personality of its own.

One may object that this is a lot of work. True, unfortunately, for the secret of good recording quality has always been meticulous care, and we see no chance of the future remedying the situation.

Reference

1. C. J. LeBel, Properties of the Dulled Lacquer Cutting Stylus, *J. Acous. Soc. Am.*, Vol. 13, No. 3, pp 265—273, January 1942.

Report on 1948 Convention of Speech Assn. of America

THE problems and limitations of bone conduction audiometry were discussed at the 1948 convention of the American Speech and Hearing Association by the well-known authority on hearing measurement, Dr. Scott Reger of the State University of Iowa. His paper, "Factors Influencing the Accuracy and Interpretation of Bone Conduction Hearing Tests," was presented in Washington, D. C., on Dec. 28th.

He first touched on the inherent inaccuracy of measurement of bone conduction and air conduction hearing thresholds by the use of tuning forks as described by Weber, Rinne, and Schwabach. These methods would be valid only if controlled by elaborate instrumentation. While the modern audiometer is superior to the tuning fork, he criticized the naive assumption that the audiometer is a precision instrument. Measurements made with a perfect audiometer under ideal test conditions are only as accurate as the audiologist's testing skill and knowledge about the limitations of his instrument and testing technique.

It is necessary to hold the audiometer's bone conduction vibrator against the skull with constant pressure, because the pressure governs the mechanical coupling between vibrator and head. Also, with some types of magnetic vibrators, the distance between armature and pole pieces (and hence the output) is affected by pressure. He used a pressure of 1000 grams, controlled by a special headband structure.

To measure the resulting vibrations of the skull, he used a crystal-type contact microphone held against the frontal eminence of the forehead by a heavy rubber band. This made possible a study of the vibration waveform, and of the vibrator linearity.

In testing a vibrator it is necessary to determine the difference between the bone conduction threshold of a given unit, and the air conduction threshold produced by the same vibrator due to leakage. The differential should be at least 15 db.

He then discussed the application of bone and air conduction measurements to the localization of ear pathology. This was then used to explain why some untrained individuals have acceptable speech even though hard-of-hearing from childhood, while others do not. Children with middle-ear impairment may hear their own voices very well by bone conduction, even though their air conduction hearing of other people is inadequate. They will tend to have normal speech development, while those with an inner ear lesion, and hence impaired bone conduction, from childhood may be in serious need of speech training. Except in extreme cases, this will be of help.

Recording Equipment

A symposium of interest to recording equipment designers was held on the morning of December 30, presenting the following subjects and speakers.

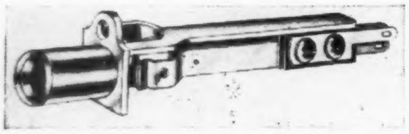
[*Continued on page 36*]

NEW PRODUCTS

NEW JACK

A new, improved ADC jack for jack panels and numerous similar uses is being produced and distributed by Audio Development Co., 2833 Thirteenth Ave. So., Minneapolis 7, Minn.

The frame of the new ADC jack is made of nickel plated, heavy gauge steel,



and is die-formed and press-welded for utmost rigidity and dimensional accuracy. The brass sleeve is nickel plated. To meet the high corrosion resistance requirements called for, silver alloy contacts and nickel silver springs were specified in the new design.

Dimensions of the new jack are standard, and like earlier types of ADC jacks it is interchangeable with any standard telephone type jack using a ¼" plug.

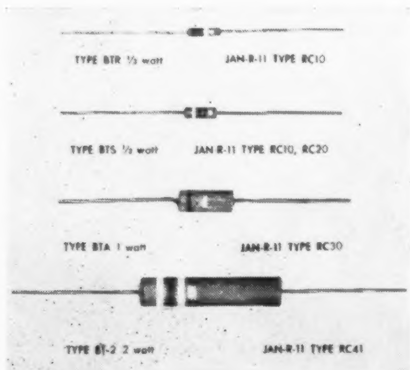
ALL-TRIODE AMPLIFIER

A high-fidelity, medium power, all-triode amplifier is announced by the Browning Laboratories, Inc., of Winchester, Mass. as a companion unit to their recently released RJ-20 high-fidelity FM-AM Tuner. Designated as Model AA-20, the new amplifier features all triode voltage gain and power stages for response within 1 db. from 10 to 17,000 cycles with less than 1½% harmonic distortion at 14 of the rated 15 watts output. Hum level is 65 db below maximum rated output. Extremely high quality pushpull input and output transformers are used and voice coils from 1.2 to 30 ohms can be matched by tap selection. The output stage is pushpull 6B4G's driven by two triode sections of a 6SN7 in cascade with separate bias rectifier. Convenience receptacle provides voltages for operating noise suppressors or preamplifiers for reluctance pickups. Use of the AA-20 is recommended with the Model RJ-20 Tuner for high fidelity installations.

NEW RESISTOR

● A positive advance in the field of electrical resistance has been established by the Engineering and Research Departments of International Resistance Co. After years of wartime and postwar exploration and development, a new insulated fixed composition resistor has been produced that challenges performance standards for that type resistance unit.

This new resistor, designated by IRC as the Advanced Type BT, is being produced in 1/3, 1/2, 1 and 2 watt ratings —equivalent to JAN Types RC10, RC20, RC30 and RC41. It meets joint Army-Navy requirements under JAN-R-11 specifications.



Production experience by IRC on this advanced resistor currently totals over 70 million units.

This family of tiny resistors (type BTR resistor body measures 3/32" x 13/32") is furnished in ± 5%, 10% 20% tolerances in RMA resistance ranges. Temperature coefficient varies from 0.02% per °C. for low ranges to 0.14% per °C. for high ranges. Depending on the size of the resistor, voltage coefficient varies between 0.0% and 0.27% per volt. Noise level is inherently uniform and low. The element, constructed to IRC's filament principle, is housed in phenolic resin; High pressure molding of the housing provides maximum security against humidity damage and moisture penetration, and enables this new resistor to withstand the most severe salt water immersion tests.

For short periods, overloads of 50% to 100% may be applied without damage to the Advanced BT, and 5 second overloads of 2.5 times rated load result in only negligible resistance changes. Frequency characteristics and other performance data are given in Technical Bulletin B-1. Copies may be obtained from International Resistance Co., 401 N. Broad St., Phila. 8, Pa.

EQUALIZER-AMPLIFIER

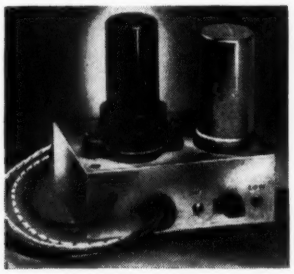
A new Equalizer-Amplifier, the Model EA-3, for use in conjunction with Astatic Corporation's highly touted Magneto-Induction Pickup Cartridge, brings to a total of three such accessory units produced by this pioneer Conneaut, Ohio, sound equipment manufacturer.

Astatic invaded less than a year ago the magnetic type cartridge field, with announcement of its revolutionary Magneto-Induction Pickup, after years of adhering solely to the development and production of crystal devices. Radical reversal of engineering precedent and drastic simplification, embodied in the Magneto-Induction unit, eliminate need for delicate handling and common sources of trouble with magnetic type cartridges, the firm

PROFESSIONAL DIRECTORY

claims. The result is peak fidelity of reproduction that is not diminished by consistent service or adverse climatic conditions.

With the advent of the new long-playing records, the Magneto-Induction Cartridge was one of the first adapted by Astatic to the new requirements. It is



thus available in models for both standard 78 RPM and long-playing microgroove platters.

The Magneto-Induction Pickup has enjoyed steadily mounting acceptance since its introduction. It is felt that availability of a third type of Equalizer-Amplifier, which lends itself most advantageously to many applications, will stimulate still further the rate with which the cartridge itself is being adopted by the trade.

WIDE-RANGE CAPACITANCE TEST BRIDGE

The General Radio Type 1611-A Capacitance Test Bridge measures capacitance over the extremely wide range of 1 $\mu\mu$f to 10,000 μf, a total spread of 10 billion to one. Over this entire range an accuracy of $\pm(1\% + 1\ \mu\mu\text{f})$ is maintained. The dissipation factor range is 0 to 60%. The frequency of the test voltage is 60 cycles.

The bridge is useful for measuring all types of capacitors, and the dielectric constant and dissipation factor of both solid and liquid insulating materials. It is also suitable for the shop testing of bushings and insulators in the electric power industry. Facilities are provided for introducing a polarizing voltage for the measurement of electrolytic capacitors.

A feature of the bridge is a unique zero-compensating circuit that balances out the initial capacitance and dissipation factor at zero setting of the dials.

The bridge is completely self-contained, including visual null detector, and operates from the 60-cycle power line. The case is of the airplane-luggage type, with handle for carrying. Over-all dimensions are 14½ x 16 x 10 inches, and net weight is 30½ pounds.

BROADCAST EQUIPMENT BROCHURE

Three new brochures describing RCA's finest AM broadcast transmitter and new AM-FM and television studio audio equipment are now available to those requesting them on broadcast station letterhead addressed to RCA field offices or to the Broadcast Equipment Section of RCA Victor, Camden, N. J., it has been announced by the RCA Engineering Products Department.

AM BROADCAST TRANSMITTER (Form 2J-4367) fully describes the BTA-50F1, latest model in RCA's series of 50,000-watt AM broadcast transmitters. The new power-saving triodes and other design features which can cut broadcasting costs up to $12,000 a year are presented in detail in the 24-page booklet.

BROADCAST TWO-STUDIO CONSOLETTE (Form 2J-4604) provides detailed information on the broadcast two-studio

consolette (Type 76C). This 20-page brochure furnishes complete operating data and specifications, as well as simplified line drawings.

CONSOLETTE SWITCHING SYSTEMS (Form 2J-4622) presents 16 pages of complete information on the latest consolette-type switching systems, Models BCS-1A, -2A, and -3A, for AM-FM and TV networks. Elaborate or simple switching problems for controlling up to 15 studios are outlined in detail in this brochure.

SOS CINEMA CATALOG

A new 64 page catalog covering every phase of motion picture theatre operation has been issued by the S. O. S. Cinema Supply Corp. Over 100 illustrations distributed throughout the book alongside the text make it simple to order the wanted part.

Sections are devoted to equipment for auditorium, booth, drive-in theatres, electrical installations, lobby, boxoffice, photography, portable and home movies, projection lighting, 16 and 35 mm projection, public address systems, theatre sound, stage, studio, recording and laboratory.

This 1949 edition is claimed to be the most complete listing ever issued in the trade and bears out the S. O. S. tradition as the "Department Store for the Motion Picture Industry."

— *News* —

NEW AUDIO CONSULTING FIRM

Audio Research Associates, 608 Fifth Avenue, New York, has recently been organized with J. H. Beaumont as general manager and L. S. Goodfriend as chief engineer. The services furnished by the new firm include anything from changing a tube in an amplifier to the complete design of a theatrically equipped auditorium, including the lighting, sound control, acoustical treatment, and general overall problems.

The staff includes a number of leading specialists in the fields of audio facilities, the theatre, architectural acoustics, recording and reproducing systems, hearing aids, residence radio systems, psycho-acoustics, patents, technical manuals, and psychological problems. Their work includes consulting, design, preparation of specifications, fabrication, installation, and operation.

* * *

ERRATA

George W. Curran, whose excellent article on the "Use of the Transmission Measuring Set" appeared in the February issue, has mentioned a few errors in typography, all on page 28. The first word on the page should be "Choose" instead of "constant." Paragraph *f* in the second column should read "The gain of the X-amplifier at the reference frequency will then be equal to the amount it was necessary to change the GS loss in (*e*)." The second paragraph in the third column should read, "The auxiliary amplifier will not be needed if the AF oscillator can deliver a level at the output of the branch pad equal to the rated output of the X-amplifier."

RECORD REVUE

[*from page 22*]

music is concerned. And I repeat, your musician is on absolutely solid ground when he objects. After all, are you designing an electronic instrument or a musical one?) Some too-pure wave form is generated that, though perhaps highly impure technically, measured for distortion, is nevertheless too *simrle*, too fixed in its pattern for the musical ear to take as the basis for musical performance.

The type of electric organ in which sliders are pulled out to add each of the overtones of a "pure" fundamental is to some extent a victim of this too-simple pattern. The fundamentals generated by this machine are incredibly dead to any musical ear, even if something less than pure by distortion measurement. Even with a maximum of tone coloration added, via the sliders, the compound tone generated is still, to most musicians, relatively dead and unmusical. That is a widespread opinion. Hence, for the purposes of my February argument on tone color vs. harmony and counterpoint, we may consider the basic tone of this type of instrument as pure—far too pure.

* * * *

All of which leads to brief mention of an electronic keyboard instrument now under development by a well-known sound engineer who shall be nameless, which I played at length last week. Good. Remarkably alive, though it is designed for the simplest uses at a low (home instrument) price. The answer to this good musical quality would seem to be (*a*) specifically, tone color directly generated along with the fundamental, as harmonic "distortion" (or so I understand it); (*b*) more generally, a rather nice appreciation of the importance of all these semi-psychological laws for musical-sounding tone, plus a common-sense, ingenious application of them within the limited means available.

Vital items were, for instance, to my ear, the excellently contrived initial *ictus*, the "speaking" of the sound; similarly, the well-calculated decay rate when the key is lifted. Neither is *consciously* noticeable, nor should be; the net effect, though, is a sense of rightness, of appropriately musical sound. A fairly simple matter of electronic principle—except for the decision as to values; and that is where it takes what this designing outfit seems to have, a sense for music and the ability to interpret that sense directly into *mfds.* Another item I found pleasing musically was the longer time required for the keys to "speak" in the low registers, as compared to the upper, an excellent and thoroughly musical idea. The lowest tones were so realistically delayed and came in with such a fine "blat" that one would swear a heavy reed was speaking. That's the kind of thing a musician can understand.

Musical Tone — Dead or Alive

Between the misprint and the correction in a monthly magazine there's a long wait—here's a bit of verbal untangling that goes back to the February issue. What came out takes the prize for insanity among our supposedly sane contributors. Naturally all our readers keep back issues of Audio Engineering. Kindly turn to p. 39 of the February issue and look at paragraph two. Note that the second line (*"that changes inde:cribably. . ."*) also appears farther on, 21 lines up from the

bottom, where it belongs. The proper line was omitted, this one installed in its place, making for a fine Department of Utter Confusion![2] Correct reading:

"*This peculiar relationship between tone-color and harmony-melody is one of the fascinations of sound as we hear it. . . .*" etc. It was a humorous accident that this very sentence concluded with the remark that it was this kind of thing that left engineers gasping like fish out of water. Some of you must have gasped several times.

RECORD REVUE

Beethoven, Trio in D, opus 70 ("Ghost" Trio).

2 (Ed. note: Ths is what sometimes happens when a compositor reassembles type after making a correction. The editors don't get a chance to check it again.)

Busch-Serkin Trio.
Columbia MM 804 (3)

Beethoven, Violin Sonata #5 ("Spring" Sonata) opus 24.
Jascha Heifetz, Emanuel Bay.
RCA Victor DM 1283 (2)

An interesting technical example of the difference between good recording and good microphoning—this has the former but is lacking in the latter. The usual excellent tonal range applies here, with what at first might seem to be an unusually strong pre-emphasis of the high end, that is more likely (as one begins to discover in this field) a purely acoustical difference that brings the highs more strongly than usual to the mike. (But note well that whether it is actually pre-emphasis or merely a matter of acoustics, the net result is the same in the listening; with this record you will find it desirable to roll off the highs and you can ascribe your own best guess as to the reason, if you don't subscribe to mine!) On the other hand the microphone-acoustic factor is not up to par here, in common with a number of Columbia chamber recordings, probably all made at about the same period and being issued irregularly. The general effect is too dead, even for a group of only three instruments. But more unpleasant is the seeming closeness of the two stringed instruments, violin and cello, which have an overly sharp and dry sound; the piano, on the other hand, is relatively off-mike, with a good sound on its own but one that doesn't mix well with the close sharpness of the strings. In some situations this perspective effect might be very useful; unfortunately the piano is notoriously hard to blend with strings at best and this only makes the musical split here a bit wider. . . However, this is still a good enough job for any listening—I'm splitting hairs for the technical-minded. Musically, Rudolph Serkin does some extraordinarily powerful piano playing, as always, and carries this performance along towards being one of the best Beethoven items for a good while.

The "Spring" Sonata, a similar musical combination, is somewhat the reverse. Though it has good tonal range, I suspect that the above Columbias have a bit more. On the other hand, RCA's acoustics and mike pickup are brilliant, both technically and with a brilliant sound. There is a trace too much violin here and there (Heifetz being the big name) but mostly the volume balance is excellent. The perspective balance is even better and considerably superior to that of the Columbia recording—here the violin and piano blend, in the general acoustical brilliance, as well as they ever can in real life. This album is a continuation of an earlier Heifetz-Bay sonata album, DM 1254.

Mozart, Piano Sonata in F, K. 332.
Vladimir Horowitz
RCA Victor DM 1284 (2)

Kabalevsky, Piano Sonata #3.
Vladimir Horowitz
RCA Victor DM 1283 (2)

Debussy, Serenade for the Doll; Poulenc, Presto; Prokofieff, Toccata, op. 10.
Vladimir Horowitz
RCA Victor 12-0428 (1)

Highly diverse samples—they couldn't be more different—of Horowitz' astonishing facility at the piano. These are, together with numerous other Horowitz recordings, the finest piano offerings Victor has made to date, and you couldn't do better than to sample this wide range of music, all canned under similar conditions, to see what Victor has to offer in the way of piano recording technique. Always a big, concert hall sound, very live, with plenty of brilliance in the piano tone. But these are the records we will most look forward to hearing on the new 45 plastic, since hiss and scratch interfere prominently in all of the softer passages. (Victor allows them to fall to very low levels.)

The Mozart sonata is a model of good playing, making what is usually played as a finger exercise into the real music that it ought to be. Kabalevsky's sonata is a harsh and, I find, rather crude bit of semi-modernism; Horowitz plays it just as harshly as he plays the Mozart with delicacy. The Prokofieff Toccata is harsh but less so and with more good sense to it, a very strong rhythm, fiery, breathtaking performance. The Debussy is lightly humorous, misty, scratchy (i.e., the shellac is

scratchy); Poulenc's *Presto* is fast, lightweight, over almost before it gets underway.

Britten, The Rape of Lucretia. (Slightly abridged)
Peter Pears, Joan Cross, N. Lumsen, etc., Chamber Orch.
RCA Victor DM 1288 (8)

This is the newest opera sensation from this leading British composer, in a performance supervised by Britten himself, an English cast that is ideally suited to this music, so utterly different from "modern" American music. This is a Roman tragedy set in a classic style, severe, simple, with small orchestra, no big noises, beautifully restrained singing; the whole thing builds by understatement and atmosphere that gets over even on non-long-playing records and is, I find, very moving and mightily impressive. Most of the words are intelligible as sung; the notes in the album give the story. The music is not at all dissonant; rather it is modal, the nearest equivalent being perhaps Vaughan-Williams.

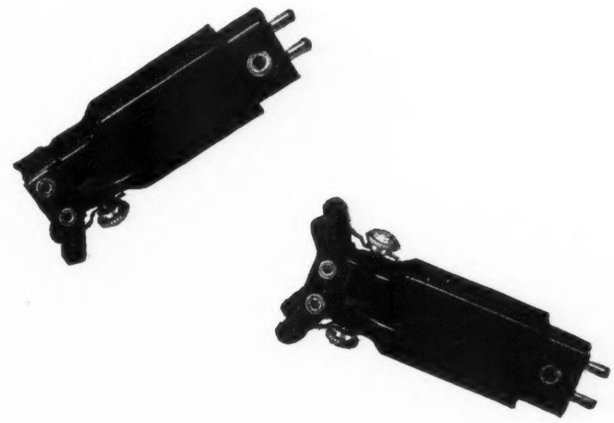
A moot point technically: was this originally one of the famous E.M.I. wide range recordings? If so, then once more, as has been noticeable before, the Victor pressings do not show it. The recording is very fine, but the highs are certainly far from exceptional and I'd say they are definitely weak. Again, it may be a matter of acoustics, but I doubt it, in view of similar effects on other Victor-pressed E.M.I. imports. Has anyone directly compared a British pressing with a Victor pressing of a recent, postwar recording? Does Victor perhaps treat the masters to a bit of polishing? If not, then what?

Mozart, Symphony #39 in E flat, K. 543.
Cleveland Orchestra, George Szell.
Haydn, Symphony #88 in G.
Philadelphia Orchestra, Ormandy
Columbia LP: 4109 (1)

Two first rate symphonies in this style, here recorded with excellent acoustics, performance and engineering; both are on the single LP record and (for my taste at least) a good combination. The two are also available in separate standard albums, 3 records to each. There is considerable difference in liveness between the two recordings; the Philadelphia job is big, brilliant, the Cleveland one somewhat less live, with a closer feeling. Impossible to know whether this is purely an engineering difference or whether it is actually a matter of musical interpretation. My guess might be that Szell, in the Mozart, purposely used a somewhat smaller orchestra as is proper, and perhaps even preferred the less live, more intimate pickup, which also is proper for the music.

Chopin, Andante Spianato and Grande Polonaise Brillante, Op. 22
Claudio Arrau, piano; Little Orch. Society, Scherman.
Columbia MX 307 (2)

This early work of Chopin's is somewhat of a freak—the first part, the *Andante Spianato,* is for piano alone, but the second part suddenly sprouts an orchestral accompaniment. Since the two parts are more or less continuous, this poses a nice recording problem, and the results here are unexpectedly happy. Most of Columbia's piano solo recording has been of the close-up, rather dead variety. Here we have a whole piece where the piano is necessarily set up in the midst of an orchestra in a largish hall or studio. I like it, and recommend that Columbia try more of the same. A nice, liquid, unpercussive tone, perfectly suited to Chopin and ac-



tually much better than the hard, brassy, triumphant Chopin that Victor's artists often give us. The Orchestra—when it enters—is not of much importance, as always with Chopin, but its background efforts set the piano off well. The *Polonaise* is mostly icing of the most glittering sort, played perfectly and just a bit icily by Arrau.

Auber, Overtures.
Boston "Pop" Orch. Fiedler.
RCA Victor DM 1274 (4)

A fine batch of rip-roaring, sentimental, dignified orchestral overtures in a real old fashioned style. A good album to have around for general test purposes, as well as casual listening. Big, pompous orchestra, nice melodies, climaxes, etc. None of it vitally important as music—but so what!

SPEECH ASSOCIATION

[*from page 28*]

Recording in the Speech Laboratory, by *Giles Wilkeson Gray*
Recording in the Speech Clinic, by *B. A. Anderson*
Recording in the Speech Classroom, by *Wayne C. Eubank*
An Engineer Looks at the Problems of Speech Recording, by *C. J. LeBel*

Professors Gray, Anderson, and Eubank emphasized the wide use of recording in speech work, and the need for high audio quality. As a teaching device, a recorder is useless unless it can exhibit a speech defect clearly to the student. A machine whose frequency range is inadequate or whose distortion is excessive cannot be used to illustrate poor diction, for example. The problem is accentuated by the fact that the speech sounds are presented alone, not in context. It is therefore highly desirable to have a frequency range of 50 to 10,000 cycles reproduced uniformly. A large proportion of the recording machines presently offered to schools for speech work are unsatisfactory. Professor Eubank noted that while tape reusability made it preferable for much routine work, it was easier, cheaper, and more compact to store material on lacquer discs.

Mr. LeBel pointed out that their complaints pertained to inexpensive home-type machines. The engineer working on inexpensive equipment has never had to achieve faithful reproduction, a moderately pleasing result being his normal goal. It is out of the question to get a 10-kc range for $135 as one of the speakers requested, for whereas a home recorder manufacturer may sell several hundred thousand units per year, a professional design will be fortunate if it sells at the rate of one thousand in the same period. Mr. LeBel showed that while magnetic recording media

had the advantage of reusability, magnetic recording equipment had maintenance problems not needed by disc equipment. These problems are head wear, head alignment, and bias variation with time.

In the resulting discussion it developed that while the schools had paid several hundred to several thousand dollars for each disc recording unit, and had no complaint about their performance, they were very unhappy about the sound quality and/or durability of home wire and tape machines. It was pointed out that they were calling for professional quality, and could expect to get it only at professional prices.

Educators may not be able to toss around electroacoustical terms as glibly as the average engineer, but they appear to have better trained ears than the average home recorder designer. It would be profitable for sales departments to drop the idea that any old piece of junk is just the thing to sell to schools. This idea was the vogue in the disc recorder field until the educators learned their lesson. Apparently we are now about midway in a similar cycle in magnetic recorders, and the disillusionment stage has already begun.

N. A. B. CONVENTION

[*from page 15*]

meetings cover training of personnel, advances in facsimile, and a report on UHF television. An FCC-Industry Roundtable is scheduled for 10:45 a.m., with Royal V. Howard of NAB as moderator.

Two interesting affairs are scheduled simultaneously for Saturday afternoon—a tour of ABC and NBC television stations, and an open meeting of the NAB Recording and Reproducing Standards Committee in which all members and interested parties are invited to participate.

RMA-IRE SPRING MEETING

● Two audio papers are scheduled for the RMA-IRE Spring Meeting, to be held at the Benjamin Franklin Hotel in Philadelphia on April 25, 26, and 27. Both are spotted for the 9:30 a.m. session Wednesday, and are:

- **Audio Power Amplifier with Positive and Negative Feedback**, John M. Miller, Jr., Bendix Radio.
- **Longitudinal Interference in Audio Circuits**, H. W. Augustadt, Bell Telephone Laboratories.

These papers should be of especial interest to the broadcast engineer, particularly those who are involved in system design.

EXPERIMENTAL ULTRASONICS

[from page 25]

Stretching The Diaphragm

The diaphragm material can be nickel, monel, German silver and so on. For any reasonable predictability of results, it must be stretched. This proved rather difficult. Welding around the rim did not work out too well, as the joint was required to be waterproof under 100 lb of pressure, and welding often left small gaps. It was finally soldered in a jig that kept the diaphragm under tension during the soldering operation. A jig was

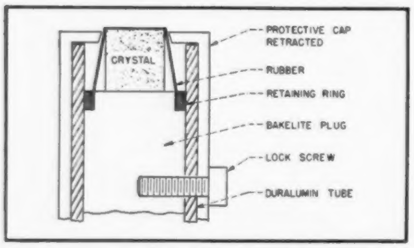


Fig. 3. Details of end of transducer probe suitable for use in air or gases.

constructed that could be placed in a furnace and brought to 400°. After precision tinning both the diaphragm and the shell, as well as the button, the assembly was slowly brought up to heat and a ring of solder flowed around the joint. Slow cooling gave us a stable assembly, and the diaphragm was turned off flush.

A diaphragm thickness between 4 and 6 mils proved practicable. The button thickness did not prove critical in the sense that a thickness of a quarter or half wave showed up in greatly increased output. The thicker the button, the less the output, in general. For stiffening, a thickness of 6 mils proved adequate.

Crystal Mounting

In general, a sliding reaction type mounting was used. If we assume the diaphragm will be bulged inward by external pressure, then we attempt to spring mount the crystal so it can move longitudinally with the slow displacement of the diaphragm. This works out well if good sliding fits are made.

Loading the crystal with a lead block did little good above 50 kc or so, a small mass sliding bakelite plug being sufficient to back the crystal.

Characteristics of Various Crystals as Mounted

In general, the Rochelle bimorph of either bender or twister type, such as are used in phonograph pickups, gives great output in isolated peaked responses, with no uniformity of output at all. Its modes of oscillation above 10 kc are so complicated as to defy analysis. If you just wish to know that some supersonic energy is in the medium, the bimorph will often be a guide with occasional flashes of output up to several megacycles, but for quantitative work it is useless.

Quartz has little response off resonance, especially when we attempt to drive one quartz with another through some medium. So in general, they are useful well above a megacycle, where other devices are unusable, and then we must frequency modulate the transmitter quartz crystal for practical applications. The combination then works rather well. With a swept band of 100 kc or so at 5 mc, and 200 volts input, the receiving unit will pick up about a millivolt of signal. Quartz must also be used when the temperature is much above 250° F.

A word of warning—practically all wartime quartz was shear cut, so do not attempt to use it under penalty of extremely complicated radiation patterns, useless for any practical purpose.

Primary ammonium tartrate crystals (PN) are about the most promising. They can be used at boiling

water temperature. They readily come up to an inch in length in pure piston action, cut with a natural period of about 50 kc, are fairly sensitive and have low noise level. They have too low a capacitance to be ideal. A half-inch cube has about 0.8 $\mu\mu f$ capacitance, so they suffer from the shunt capacitance of a long lead or a vacuum tube voltmeter or oscilloscope.

Since they very readily dissolve in water, any mounting must be waterproof. Any handling with the fingers should be avoided, as they are gradually eroded by finger sweat. They are quite soft, about like a cube of sugar in hardness, so they are easily damaged by a blow, or a sharp-pointed instrument.

Another controlling difficulty in making a stable permanent mounting of the PN is the fact that no pressure can be used except on the working faces. These come marked with a little black dot, and the crystal must be wholly supported by simple direct pressure on these faces only. Otherwise its sensitivity and resonant period will be seriously and erratically affected.

The writer has used these mostly in the form of cubes. Samples are furnished to very exact dimensions, and dead square.

When mounted in the shell of *Fig. 1,* they are very stable and uniform. Care must be taken that no sharp blow on the diaphragm crushes the crystal but a reasonable amount of care will prevent this. However, the metallic diaphragm lowers the sensitivity very much when used in air or a gas, so the mounting shown in *Fig. 3* was developed.

This mounting as a probe has several advantages. The stretched rubber of a thickness of about 0.3 thousandths damps the crystal somewhat, but an absolute minimum. It has no characteristics of its own to speak of as a diaphragm. However, it does not do a very good job of protecting the crystal, and when used as a contact microphone considerable care is required not to damage the crystal. When used on the body there is little danger, but when working around metal parts that have sharp edges, great care must be used. The sliding protective metal sleeve should be only retracted when the crystal is actually in use.

When used with a high-intensity source, such as the Hartmann generator, this unit gives out a volt or two to drive an oscilloscope, if within two inches of the generator, and in the maximum field. It will give the same with the parabolic reflector ten feet away, sharply focused on the face of

the crystal. However, this is only true with about two leads about two feet long and separated. If a six-foot concentric cable is used the signal drops rapidly for each foot of cable. For serious work a cathode follower stage right at the crystal and impedance matching itno the 52-ohm line is required.

Unit Used as Transmitter

Any PN crystal will take 200 volts of drive. If you wish to operate over a wide band of frequencies this offers some problems.

Suitable oscillators should cover the spectrum from ten to a few hundred kc or higher. The output must, of course, be stepped up in voltage by some means.

One means is a tuned circuit right at the crystal. Due to its very small capacitance, this is rather easy to do. Somewhat more difficult is designing a wide-band transformer to step up the 52-ohm line to match the crystal. Another answer is a driver tube at the crystal, choke-coupled to it.

A warning on using the crystal as a self-frequency determining oscillator. By the time the crystal is loaded with a diaphragm and damped by the mounting, it is seldom possible to have it develop sufficient reactance to act as a normal oscillating crystal. This is especially true if it is immersed in water, where the Q often falls to ten or less. The writer made a determined but unsuccessful effort to lock such a crystal to an unstable oscillator.

Working Units in Pairs

Two similar units work well in either air or water. For a first experiment, have them face to face and join them with a drop of water. The signal transmission will be very good.

If they are used with only air in between, very high values of standing-wave ratios will be noted as they are moved apart. These may be 40 db or so. These are obtained only when they are dead parallel and dead concentric on the same axis.

If used in a test chamber, as shown in photograph, some interesting effects can be observed. If we gradually fill the chamber with water, reflection from the underside of the water will be almost 100% and wave cancellation will occur, and by careful filling drop by drop a range of signals of 60 db or more will be had. This means that a test cell such as this must be a complete housing with no water surface on top. So the filling orifice must be closed by a plug whose inner wall is flush with the inner wall of the tube. With a suitable selective receiver, almost any of these units

will pick up the ticking of a watch up to 200 kc or so, even through several hundred feet of wire. Some interesting work might be done on machinery n o i s e, especially high-speed machinery. Most of the analyzing now done acoustically cuts off at rather low frequencies.

It is somewhat of a shock to find that the ordinary telephone receiver, when overloaded by loud speech, gives out detectable energy up to 100 kc at least. We might investigate direct pickup of noise above 20 kc from a phonograph needle tip. The noise spectrum direct f r o m a grinding wheel may tell us how many of the cutting particles are engaging the work, and might be used as a feed control. We know it will be difficult for a reader of AUDIO ENGINEERING to locate a cockroach to chivvy with ultrasonics, but if he is given his dinner on the diaphragm of one of these units the noises are very unusual.

In the next article of this series we will discuss the design of suitable receivers, tuned voltmeters, and the like. It is obvious to select a communications type receiver for frequencies above 550 kc or so, but many of them are too sharply selective for some uses. The range from 10 kc to 550 kc requires very special design indeed.

MEASURING PROCEDURES FOR MAGNETIC RECORDING

[*from page 19*]

um, the magnetic heads, the bias level, the signal level, the amplifiers, and the speed of the medium. The last four items can be specified within reasonable limits. However, it seems to be impossible at the present time to specify a standard medium, or standard recording and reproducing heads. This being the case, the frequency responses of different media may only be compared directly when these responses are made using the same or exactly equivalent magnetic heads. Needless to say, the testing procedure must follow certain standard specifications. It is the purpose of this section to state and explain these standard specifications.

I. Speed of the Medium

a. *Magnetic Wire.* For magnetic recording wire, the speed of the medium for frequency-response measurements shall be two feet per second ± 2%.

b. *Magnetic Tape.* For magnetic coated recording tape, the speed of the medium for frequency-response measurements shall be 7.5 inches per second ± 2%.

II. Signal Level

The recording signal level shall be set at the *Standard Recording Level.* In a given magnetic recording system, the standard recording level is the value of audio current in the recording head such that the resulting remanent induction in the recording medium produces an open-circuit voltage at the terminals of the reproducing head which is approximately 12 db. lower

than the *Saturated Signal Output*, provided that the frequency is the *Frequency of Maximum Response* and that the bias is set at the value of *Operating Bias Current*.

Saturated Signal Output

The saturated signal output is the maximum voltage which appears at the terminal of the reproducing head when the field intensity in the gap of the recording head is of such magnitude that a further increase does not result in an increased reproduced signal measured at the frequency of maximum response and at the operating bias current.

Peak Recording Level

The peak recording level shall be that value of input which results in 10% total harmonic distortion at 400 cycles, as determined in accordance with distortion measurement procedure.

Operating Recording Level

The operating recording level shall be such that modulation peaks do not exceed the peak recording level.

Frequency of Maximum Response

The frequency of maximum response is that frequency of recorded magnetic signal which produces a maximum open-circuit output voltage at the terminals of the reproducing head when a constant-current recording signal at the standard recording level is used in the recording head.

III. Operating Bias Current

The operating bias current shall be that value which results in maximum output at a frequency of 200 cycles at the standard recording level. The exact value shall be the algebraic mean of a higher and lower biasing current which results in a 1 db. decrease from the output of maximum response.

IV. Amplifiers

a. *Recording Amplifier.* The recording amplifier shall be designed and connected to the recording head in such a manner that the signal current in the head for constant voltage supplied to the amplifier input does not vary more than ± 1 db. over any specified useful bandwidth of the recording system.

b. *Playback or Reproducing Amplifier.* The frequency response of the playback amplifier shall be flat in terms of voltage output within ± 1 db. over any specified useful bandwidth of the recording system.

It should be pointed out that when high-impedance magnetic heads are used, resonant effects may give an erroneous impression as to the unequalized frequency response of a magnetic recording system. In the recording head, the resonant effect may result in a gap field strength which is considerably higher at the resonant frequency than at other frequencies. In the playback head, the resonant effect may account for as much as 6 or 8 db. higher response at the resonant frequency. This may be a desirable method of obtaining post-emphasis characteristics of a magnetic recording system but does not tend to improve the signal-to-noise ratio of the system.

In making frequency-response measurements of the medium, it is therefore recommended that

(a) Signal level be that specified as Standard Recording Level;
(b) Bias be that specified as Operating Bias Current;
(c) Amplifiers meet the requirements as specified in Section IV;
(d) Measurements should extend either side of the frequency of maximum response until noise becomes the

limiting factor or the output is 20 db below that obtained at the frequency of maximum response.

In making overall frequency-response measurements of a magnetic recording system, the same conditions as above shall apply, except the amplifiers shall be those used in the system.

2. DISTORTION

In measuring the distortion of a magnetic system, or any system where a moving medium is involved, certain precautions must be observed in order to obtain a true distortion evaluation. Variations in speed of the medium may cause frequency and phase shifts which exceed the limitations of the measuring equipment. This is especially true if the equipment is of the type where the fundamental is filtered out by means of a sharply tuned filter and the residue measured, or where the individual harmonics are selected and transmitted through a band-pass filter of only a few cycles in width. It is believed that the band pass of the filter should be broad enough to permit frequency changes of the order of at least plus and minus 5 percent. If harmonic distortion measurements at only one frequency are contemplated, it is suggested that 400 cycles be chosen as the frequency, since distortion measuring equipment which fulfills the requirements stated above exists.

Harmonics above the fifth may not be too significant, and if such is the case, the playback system need only be compensated so that the overall response is flat within one db from at least 400 to 2000 cycles. When making distortion measurements, the recording current should be held constant as recommended for frequency-response measurements. In order to minimize the high-frequency noise, so that is is not an appreciable part of the distortion reading, it may be desirable to permit the high-frequency end to roll off above 2000 cycles.

Obviously distortion measurements may be made at any recording level, and many measurements should be made at different recording levels and with various amounts of biasing current in studying the characteristics of the system. Such a study permits a more accurate determination of the normal recording level and operating bias.

The recording level is usually not too important from the viewpoint of the audio power, as the actual power required is very low and more power, if needed, can easily be obtained. Output voltage from the magnetic head, on the other hand, is low and requires considerable amplification, so that it may be of greater significance when comparing media. It is, therefore, suggested that when making distortion measurements both input and output levels be given.

It is therefore recommended that total harmonic distortion measurements be made

(a) At 400 cycles (measurements at other frequencies may be made and so indicated, but figures of the 400-cycle distortion values should always be included);
(b) With a system which is flat within one db from 400 to at least 2000 cycles;
(c) Using the value of biasing current as defined by Operating Bias Current;
(d) With inputs 6 and 12 db below the peak recording level.

3. NOISE

a. *System*

In order to specify noise in a manner that shows the capability of the equipment, two measurements are recommended: First, a "high-frequency" measure-

ment which excludes hum and other low-frequency noises and evaluates the effectiveness of the Erase, the result of the addition of Bias (used during recording), and the high-frequency noise present in the system; the second method includes the entire frequency spectrum covered by the system and is therefore an "over-all" measurement.

For the high-frequency measurement, a 250-cycle high-pass filter should be used to minimize hum and other low-frequency noises. The equipment noise, with the medium not running, should be at least 10 db below the value measured with the medium moving, in order to obtain reasonable accuracy of system noise measurements. If this is not the case, both measurements, "equipment" and "system high frequency," should be stated.

Since the measurement is usually one of comparison, it is recommended that the comparison be with the reproduced output obtained with the peak recording level as defined previously, and the noise be stated as so many db below this value. The frequency range covered by the system is significant in noise measurements, and it is recommended that the playback system be at least equalized flat within one db between 200 and 2000 cycles. The specified range should be expressed as extending to the frequency at which the output drops 15 db below the flat response portion. For high-frequency noise measurements, the 250-cycle high-pass filter will, of course, determine the low-frequency cutoff, and the range would be specified as being from 250 cycles to some frequency where the output is 15 db below the flat response portion.

For an over-all noise measurement of the system which includes low-frequency noise, such as hum, the 250-cycle high-pass filter should be removed and the noise measured as before. The frequency range should be stated as extending from some low frequency, which is 15 db below the flat response level, to a high frequency, which is also 15 db down.

It should be noted that noise measurements require about the same equalization as distortion measurements, and it may, therefore, be possible to use the same equipment by including a 250-cycle high-pass filter, since the distortion meter of the type available is also designed for noise measurements.

In making both the overall and high-frequency noise measurements, the medium should be erased and bias applied as normally done in recording, but without signal.

It is therefore recommended that when making noise measurements pertaining to the system

(a) The medium should be erased and bias applied as is normally done in recording.
(b) The playback system be compensated flat within one db from 200 to at least 2000 cycles.
(c) The reference signal be 400 cycles recorded at the peak recording level and the noise expressed in db below this level.
(d) A high-pass filter cutting off frequencies below 250 cycles be used.
(e) The frequency range be stated as from 250 cycles to the frequency at which the response is down 15 db from the flat response portion.

Note: If the equipment noise (medium not moving) is not 10 db or more below the system noise, both values should be stated. When making noise measurements of the overall system, which would include "hum," the procedure outlined for system

noise should be followed, except that

(a) The 250-cycle high-pass filter not be used.

(b) The frequency range be stated as extending from the low frequency which is 15 db below the flat response level to the high frequency which is also 15 db below.

b. *Medium*

Where a noise measurement which properly evaluates the capabilities of a medium is wanted, great care must be taken in erasing the medium prior to the noise measurements. It has been found that asymmetry of wave shape of erase and bias of the high-frequency supplier affects the resulting noise. Therefore, to avoid possible errors due to such an effect, it is recommended complete erasure be obtained by the use of a strong low-frequency (60-cycle) magnetic field (solenoid coil structure), having gradual decreasing strength where the medium leaves the field. Neither high-frequency erase or biasing fields should be applied after the low-frequency erase, and every effort should be made to completely demagnetize the reproducing head or any other magnetic material which may be in contact (or sufficiently close to cause magnetization of the medium), before measurement. The reference signal of 400 cycles should be recorded on a separate section of the medium or the reference level established immediately prior to the low-frequency erase. Since equipment hum should not be chargeable to medium noise, a 250-cycle high-pass filter should be used during measurements.

It is therefore recommended that when making noise measurements pertaining to the medium

(a) The playback system be compensated flat within one db from 200 to at least 2000 cycles.

(b) The reference signal of 400 cycles be recorded at the peak recording level and the noise expressed in db below this value.

(c) A high-pass filter cutting off frequencies below 250 cycles be used.

(d) The medium be erased completely, using a strong low-frequency field if necessary.

(e) The frequency range be stated as from the cutoff of the filter, 250 cycles, to the frequency at which the response is down 15 db from the flat response portion.

Note: If the equipment noise (medium not moving) is not 10 db or more below the system noise (medium moving), high values of noise should be given.

Submitted by:
Dr. S. J. Begun, Chairman
Mr. L. C. Holmes
Mr. H. E. Roys

DISC RECORDING

[*from page 13*]

ful copy. Also, when making recordings purposely for dubbing, lower levels are used in order to take advantage of the reduced cutter distortion, and care is taken to work at the outside radii as much as possible. By careful attention to response and distortion, it has proven possible to rerecord speech program as many as five times with very little deterioration of quality. The accuracy of the complementary NAB pre-emphasis and de-emphasis networks allows the faithful

preservation of the original response of any phonograph record or transcription recorded at outside sources. Good advantage is taken of this factor when dubbing flip sides of phongraph albums used in daily symphony programs. To keep the radius losses at 78 rpm from doubling during this procedure, additional equalization is available during the last minute of travel by moving the radius-equalizer control switch from "off" to "low-fixed." By addition of the "off," "low-fixed" and "high-fixed" positions to the three basic curves, nine curves are available.

Although such a system of equalization is not applicable to all conditions of recording, it has proven remarkably practical for speech, for all network recording, and for most music. The tracing distortion generated by this system does not become severe until the last equalization step is reached, and even that is tolerable when played with a good magnetic pickup capable of rejection o fall vertical components. The segues and overlaps of the sections of a half-hour show are unnoticeable even on better-than-average home radios.

Photographs by Rueben Lawson, Jr.

— Letters —

[*from page* 6]

as a reader of at least 13 radio magazines every month, I can testify that he is the first to do it.

The Columbia-Victor controversy is so confused and befuddled by corporate names, reputations, advertising, and quasi-engineering considerations that just about everyone in the technical world has got to the point of holding his head in his hands trying to stave off apoplexy. Magazines and newspapers have skirted the subject, adding to the confusion by playing up the befuddlement of all concerned. Canby's comments are the first honest, objective, intelligent ones to appear.

The comments are objective because —paradoxically—they are subjective. Everyone has so far forgotten that the only purpose of any record is to enable someone to listen to music! And music is designed for and produces purely *subjective* reactions. The only objective view that can be valid must be based on a subjective reaction.

The tremendous, unbelievable, vital fact about LP's is that you can actually forget you are listening to a record! For someone who has been a record addict for years, elimination of the high-pitched surface hiss, the mental picture of the needle travers-

ing a fast-moving disc, and all the other nerve-shattering disadvantages Mr. Canby mentions is like finding a diamond on the street. You can listen to the music—and not worry. I have bought about 15 LP's since September and will never buy another 78. I have even duplicated some of the music I already had on 78's and the difference makes the LP music sound and feel like a new, unfamiliar composition.

No matter how good Victor's disc sounds (and they do sound good—I've tried them) no one can convince me that Beethoven scored a pause in the orchestration every 4½ minutes. As long as the pause is there, it isn't Beethoven—it's a "canned" imitation.

I regret to say that I don't like the new cover design as well as the old. The magazine, however, is definitely keeping up its standard.

Richard H. Dorf
255 W. 48th St.,
New York City

PRODUCTION TAPE RECORDING

[*from page 21*]

Details concerning the manufacture of additional such machines have not been completed.

With the new machine, tape can be recorded with a single magnetic pattern in the center, or with a double pattern of two magnetic paths side-by-side on the tape.

One path plays as the reel unwinds forward, the other path functioning when the tape reverses, which is accomplished automatically in a fraction of a second. The double pattern affords twice the playing time with the same amount of tape.

The new multiple recording machine is designed so that it can record either the single or double pattern type of tape. It can record both paths on the double pattern tape simultaneously. In addition, it can be adjusted by switch control for recording different length reels, and for different speeds.

Reels having 600 feet of tape, double pattern, and a playing speed of 3¾ inches per second can be turned out at the rate of 48 per hour, each reel having a full hour's playing time.

Reels with 1,200 feet of tape, double pattern, and a playing speed of 7½ inches per second can be turned out at the rate of 32 per hour, each reel having one hour of transcribed material.

In addition, 1,200-foot tape reels designed for playing speeds of 15 and 30 inches per second can be produced, it was announced. Master transcriptions from which the tape records are made, can be played at varying speeds, to fit the requirements of the job.

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